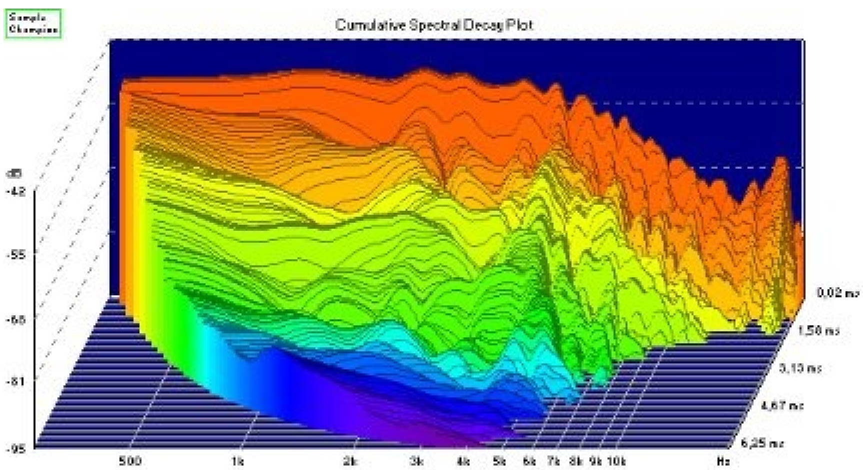
Sample Champion PRO 3.8

User Guide



by Paolo Guidorzi, PureBits.com

Contents

Welcome to Sample Champion!

Sample Champion is a powerful real-time dual channel MLS (Maximum Length Sequence) analyzer. We recommend a careful reading of this manual before using the program.

For a survey of the program, read **General Overview**, then go to **Quick Starting Guide**.

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File Management

Measurement types

Settings

Quick Settings

Tools

Other features (available in the plugins)

BEFORE STARTING...

READ CAREFULLY: To prevent any resizing or reallocating of the virtual memory during measurements, Sample Champion allocates, once started, **ALL** RAM memory and system resources required for working.

This software requires a substantial amount of RAM memory and system resources.

This is a consequence of the very high accuracy used for all internal operations. For example the FFT (limited for the user to 64K, but extended for some internal operations to 256K) is computed on double-precision complex data and using both frequency bands (positive and negative frequencies) to perform some specific signal processing operations.

IF error messages such as:
- Plugin_default not found
- Canvas does not allow drawing
- Uncorrect parameter
...
appear, most probably Sample Champion does not have enough system resources available for running correctly.

Solution:

Leave at least 2-300 MB of free space on the disk where the Windows swap file is allocated (usually disk c).

Close **any** other software before starting Sample Champion (this is always recommended).

Disable all running programs and system tasks. Check the active running tasks pressing once CTRL-ALT-DEL.

Tip: click on the windows taskbar **Start / Run** and type **msconfig** then in the Automatic Start section uncheck all the useless tasks and reboot the system.

Ideal system configuration:

- pressing CTRL-ALT-DEL only **Explorer** is shown.
- the system tray (right bottom corner of windows bar) is **empty** (only the clock is shown).

General Overview

Sample Champion can operate in real-time (while performing a measurement) or in post-processing mode (at the end of a measurement cycle) or with loaded files. There are no differences between these operating modes.

Basically, there are different types of windows, each with different contents and behaviors:

- **Time domain windows**
  - Scope windows
  - Impulse Response windows
- **Frequency domain windows**
  - FFT of *Selection in scope window*
  - FFT of *Selection A in impulse response window*
  - FFT of *Selection B in impulse response window*
- **Plugin windows**
  - Frequency domain plugins
  - Time domain plugins

All frequency domain and plugin windows have a specifical ID type (shown in the lower right hand corner of frequency windows) to distinguish them. This ID is written also inside the ***.SPE*** (frequency) files.

Sample Champion uses some memory banks to store Time and Frequency data. To reduce RAM memory requirements, only one memory bank of each type is used at the same time.

For example the user can view simultaneously:

1 *Scope* memory bank
1 *Impulse Response* memory bank
1 *FFT of scope selection* memory bank
1 *FFT of impulse response selection A* memory bank
1 *FFT of impulse response selection B* memory bank
1 *Frequency domain plugin* memory bank

For each memory bank, it is possible to open as many view windows as desired. For example, the user can open simultaneously two impulse response windows showing different time intervals or with different time scales (X axis zoom). The data of the same frequency bank can be plotted in different windows with different Y axis view options such as Magnitude linear, Magnitude (dB), third-band of octave, phase...

You can find some examples in the **Quick Starting Guide**.

System Requirements

Hardware:

Standard PC, Pentium or later, or compatible processor
(Pentium II 200MHz or faster, or equivalent compatible processor, recommended)
64 MB RAM (recommended: 128 MB or more)
Fast video card (recommended settings: 64K colors or greater, 800x600 or greater screen size)
Windows compatible full-duplex 16 or 24 bit sound card
10 MB Hard Disk space, at least 200 MB Hard Disk free (for Windows swap file)

Software:

Microsoft Windows 95, 98, ME, 2000 or XP
Full-duplex sound card drivers (supplied by sound card manufacturer). Only standard MMSYSTEM support is required (MME / WDM drivers).
DirectX or ASIO support is not required.

Quick Starting Guide

Here you'll find a step-by-step procedure to start using Sample Champion

It is assumed that a full-duplex sound card is installed and working.

In order to make some tests, it could be a good idea to have a **MICROPHONE** (connected to the MIC IN of the sound card) placed in front of a **LOUDSPEAKER** (connected to the output of the sound card, using a power amplifier, if necessary).

Alternately, a short jack-jack cable (or as required by the sound card) could connect direcly the **LINE OUTPUT** of the sound card with its **LINE INPUT** (loop back).

Now we are ready to start some measures!

Start Sample Champion

Open the settings window and set the following parameters

General Settings:
- INPUT: Mono, 4K
- OUTPUT: 16 bit, Mono
- Disable Automatic Time Latency Calibration
- Sampling rate: 48000 Hz
- Block: 1
- Mode: Repeat
- Step: 2
- MLS Type: 4K MLS (Taps 12,7,4,3)
- Level: 50%

Advanced Settings:
- Uncheck everything
- Check only "Remove offset from sampled data"

FFT Settings:
- FFT Size: 4K
- Weighting window: Blackman-Harris, Half-right
- Check "Show window..."
- Uncheck "Phase shifting..."
- Check "Transfer Function Measurement"
- Write a voltage value in the "Signal Amplitude" field, for example 1000 or 2000 (milliVolt)
- Select Channel 1(L) for all three cursor selection channel assignements

Compensation Settings:
- Uncheck INPUT Correction Enable checkbox
- Uncheck OUTPUT Correction Enable checkbox
- Uncheck Pink Filter checkbox

Calibration Settings:
- Uncheck Manual or Automativ Latency Calibration
- Do not change Amplitude calibration

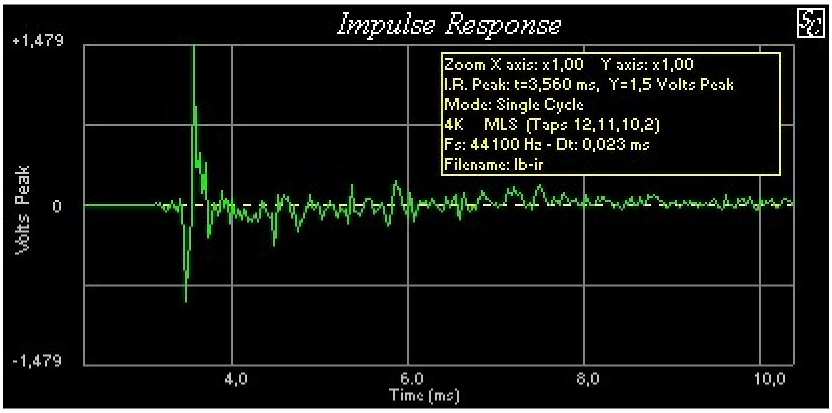
Open an Impulse Response window (press)

Press and select "Selection A in the Impulse Response Window"

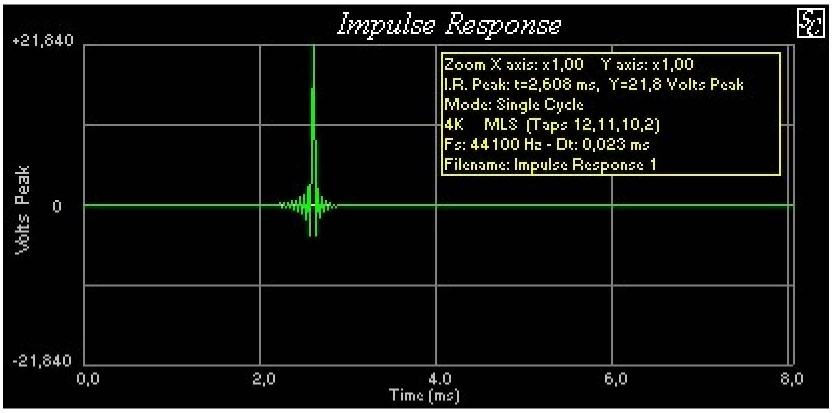
Press to have a more comfortable view (if you are using 800 x 600 pixel windows desktop setting, it could be useful to hide the taskbar to get more work space)

Now press and a single cycle MLS measurement should be performed. Click 2 times on the **Show Info on image** icon inside the Impulse Response window to see extra information.

If you are testing a loudspeaker, the Impulse Response should look like the following picture (obviously, with the Impulse Response of **your** loudspeaker!):

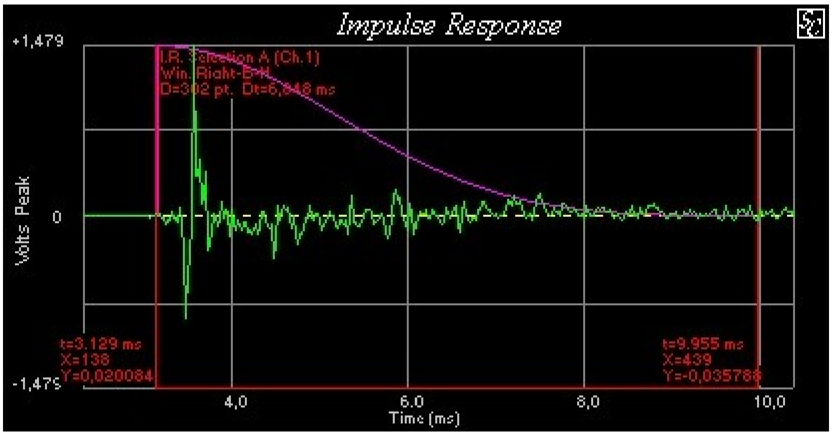


If you are using the loop-back cable, the Impulse Response should look like in the following picture:

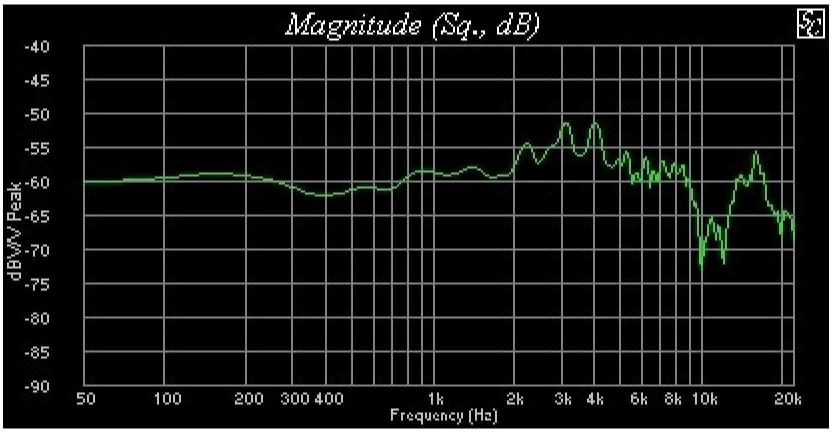


Note that for this test the latency time calibration has been manually set to zero. As side effect of the loop back measure, we have found that the sound card used in this test (a Turtle Beach Pinnacle) has a latency time of 2.6 ms (corresponding to sample number 115), so the number to write in the latency time calibration setting is "-115". But leave this setting disabled at present.
Should the latency time have been compensated, the peak of the measure of the loudspeaker would be positioned at t=0.96 (=3.56-2.6) and the loop-back measure peak would be positioned in the proximity of t=0.

To analyze the measure performed on the loudspeaker, now click on the "Make/Change Selection A" icon inside the Impulse Response window ( ). The cursor will become a cross. Click now on the I.R. image, just before the peak and, keeping the button pressed, move the cursor to the end of the Impulse. Leave then the mouse button.



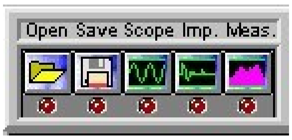
In the Frequency domain window, now, the FFT of the Selection should be plotted, in the desired viewing mode.



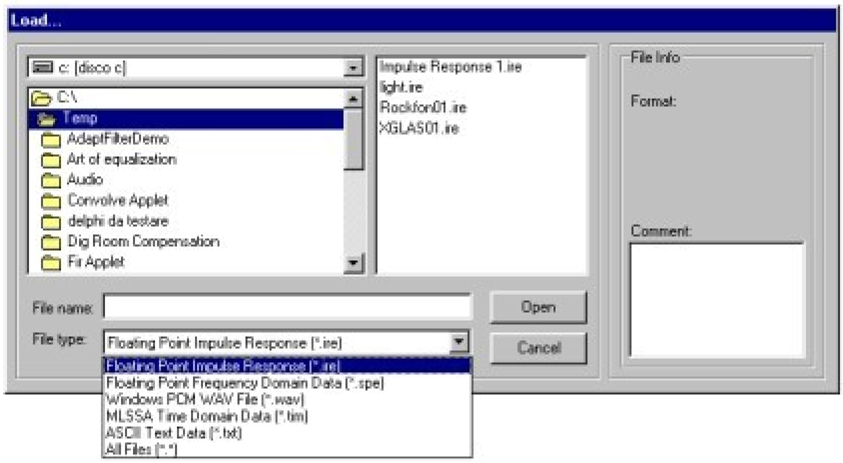
Now "Selection A" (the samples between the two red cursors) can be moved or modified using the suitable buttons. The plot in the frequency window will change correspondingly. The same procedure can be followed to analyze the content of the scope window (opening the corresponding frequency window). Now you can proceed with time and amplitude calibrations and discover all other great features of Sample Champion.

The above analysis procedure can be tested also on a loaded file. Load, for example, the demonstration file **Example1.ire**.

Open



 Open



The following formats are supported for input files:

.IRE: Impulse Response (native format, 80 bit floating point)
.SPE: Spectral data, Real + Imaginary parts (native format, 80 bit floating point)
.WAV: PCM 16, 24 and 32 bit (integer) and IEEE 32 and 64 bit (floating point)
.TIM: MLSSA™ data
.FRD: Frequency Response, ASCII format (*)
.TXT: ASCII raw exported data

- Opening a file of a certain type, the corresponding window will open (if it is not already open)

- Global sampling rate is adjusted according with the loaded file

- When a file is selected in the list, some information will be shown on the right side of the window

- Multiple views of the same file can be opened by using the corresponding button or measurement selection

– **Most Recent Used files** are added in the bottom of the File menu. If the **Show Open Dialog** option (at the end of File menu) is checked the open dialog is shown when a file in the MRU list is selected, otherwise the file is opened.

See also **Settings / File Options**

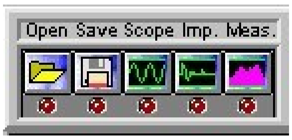
(*) FRD files are an ASCII stadard format for frequency response exchange between different software systems.

This is an FRD file format example:

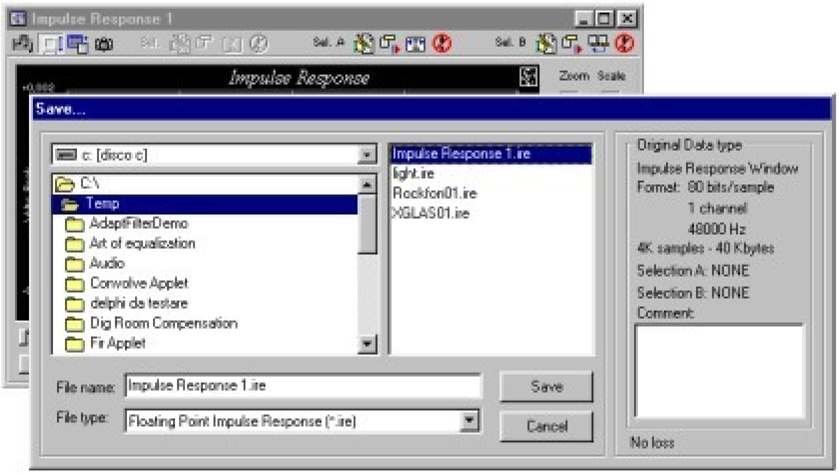
```
* Frequency Data Exported from Sample Champion
* DeltaF (Hz) = 5.85937500
* FFT Length = 8192
* Data (from f=0 to f=24000.00000000 Hz):
* FREQUENCY ---------- SPL -----------   ----------PHASE ---------
5.859375000  -6.6413279209929422  161.172205576833990
11.71875000  -6.2717003153797555  142.893539456514790
17.57812500  -5.7170485135652076  125.547620204225055
23.43750000  -5.0415376898741889  109.294398830669541
29.29687500  -4.3013566780912727  94.1163743150220921
35.15625000  -3.5376477339700959  79.8990622262171842
41.01562500  -2.7771475142395961  66.4949803316312349
46.87500000  -2.0357062995903430  53.7595379304933006
52.73437500  -1.3218866288923240  41.5660817383207405
...
```

Sample Champion can save and load frequency data also in this format. In order to have the better results, the current FFT size setting in Sample Champion must be the same of the loaded FRD file.

Save



Save



- The saving formats available depend on the current active window. For example, to save an Impulse Response, click on the Impulse Response Window and then click on the save button. Saving formats:

- If the active window is an **Impulse Response Window**:

.IRE: Impulse Response (native format, 80 bit floating point)
.WAV: PCM 16 and 32 bit (integer) and IEEE 32 and 64 bit (floating point)
.TIM: MLSSA data
.TXT: ASCII raw exported data

- If the active window is a **Scope Window**:

.WAV: PCM 16 and 32 bit (integer) and IEEE 32 and 64 bit (floating point)
.TIM: MLSSA data
.TXT: ASCII raw exported data

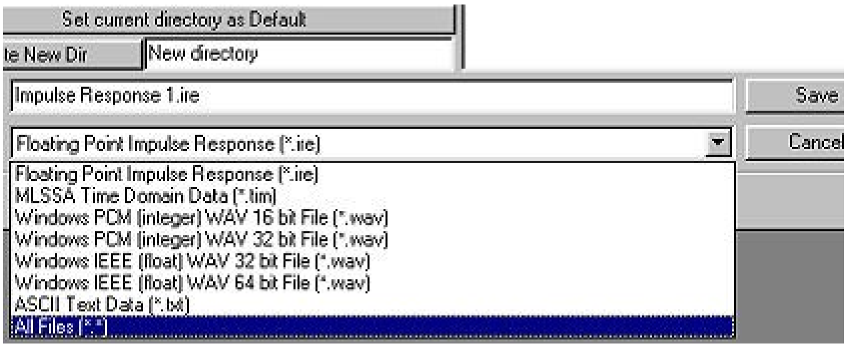
- If the active window is a **Frequency Domain Window**:

.SPE: Spectral data, Real + Imaginary parts (native format, 80 bit floating point)
.FRD: Frequency Response, ASCII format (*)
.TXT: ASCII raw exported data

- If the active window is a **Frequency Domain Plugin**:

.SPE: Spectral data, Real + Imaginary parts (native format, 80 bit floating point)
.FRD: Frequency Response, ASCII format (*)
.TXT: ASCII raw exported data

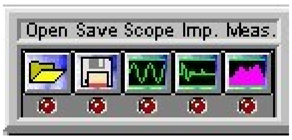
- **Most Recently Used files** are added in the bottom of the File menu. If the **Show Open Dialog** option (at the end of File menu) is checked, the open dialog is shown when a file in the MRU list is selected, otherwise the file is opened.
The WAV format type can be selected in the Save window:



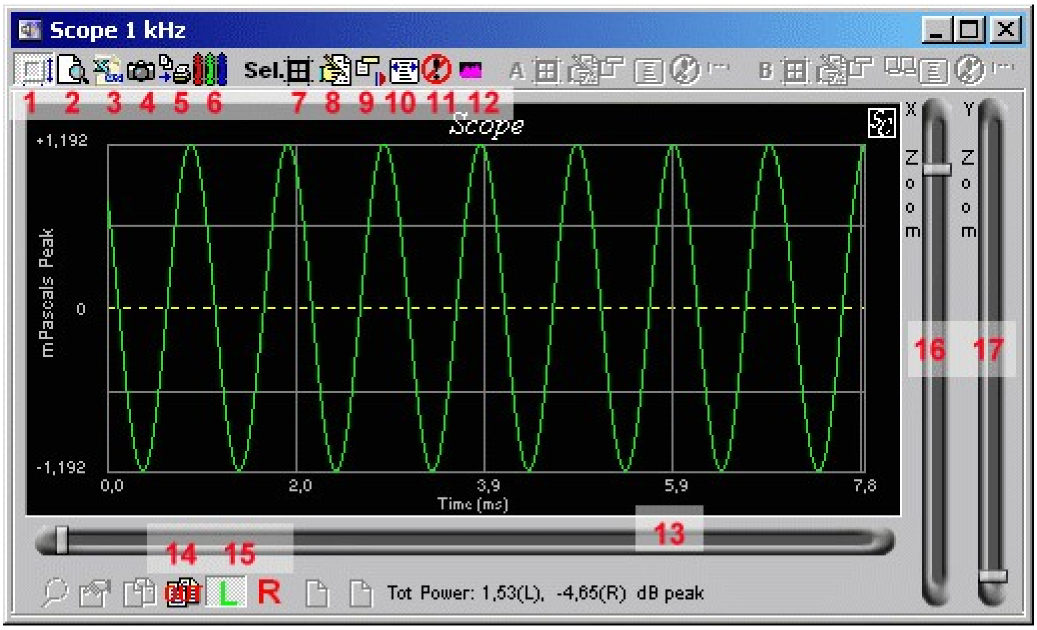
Some additional options must be taken in account when data are saved as WAV files (Settings / File Options).

See also **Settings / File Options**

Scope Window



Scope



1 - AutoScale (ON/OFF)
2 - Set View Time Interval (Lock/Unlock)
3 - View data values / Export time data as XLS (Excel) file
4 - Save snapshot of the data window as bitmap image
5 - Print the data window
6 - Set custom colors

Data Selection (for frequency analysis):
7 - Manual Data windowing and additional information
8 - START selection (if no data are selected) or CHANGE selection size (if some data are selected), moving the selection cursor nearest to the mouse pointer
9 - MOVE selection
10 - EXTEND selection to the current maximum input length or maximum FFT size
11 - DISABLE selection
12 - Quick Spectrum
13 - SET plot starting point
14 - Overlay generated Signal
15 - PLOT Left and/or Right channel data (if the measurement is stereo)
16 - ZOOM time scale
17 - ZOOM amplitude (only view)

- Pressing the **scope** button an empty window will open.
- When one or more **measurements** are performed, the sampled data will be plotted there.

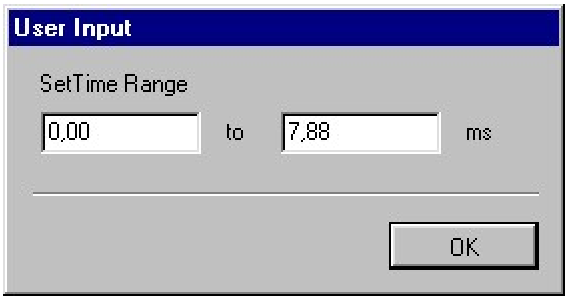
Scope Window Features:

- **AutoScale (ON/OFF) (1)**

This function allows to enable/disable **Y autoscale** for the current plot.

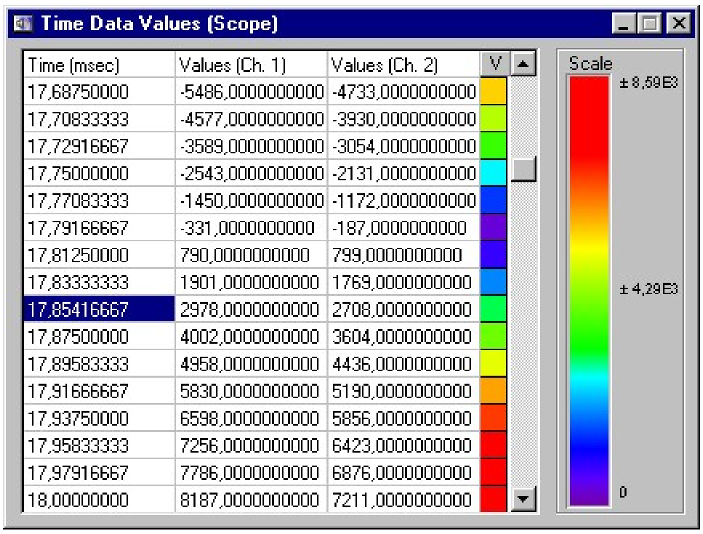
Tip: We suggest to disabled this function after the **first** measurement and enable it again (just a few moments) only if the measured signal falls outside of the plot bounds.

- **Set View Time Interval (Lock/Unlock) (2)**



This function allows to set accurately the graph time range. When it is enabled, the scroll bars in the Scope window are disabled.

- **View data values (ON/OFF) (3)**



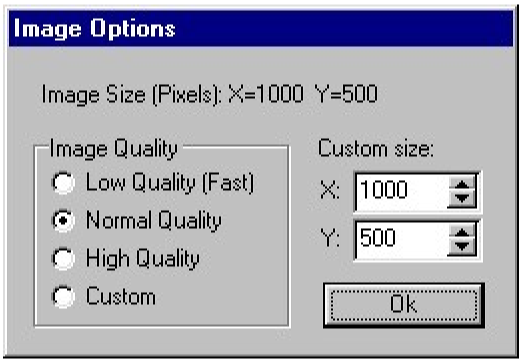
Time (msec)	Values (Ch. 1)	Values (Ch. 2)	V
17,68750000	-5486,0000000000	-4733,0000000000	
17,70833333	-4577,0000000000	-3930,0000000000	
17,72916667	-3589,0000000000	-3054,0000000000	
17,75000000	-2543,0000000000	-2131,0000000000	
17,77083333	-1450,0000000000	-1172,0000000000	
17,79166667	-331,0000000000	-187,0000000000	
17,81250000	790,0000000000	799,0000000000	
17,83333333	1901,0000000000	1769,0000000000	
17,85416667	2978,0000000000	2708,0000000000	
17,87500000	4002,0000000000	3604,0000000000	
17,89583333	4958,0000000000	4436,0000000000	
17,91666667	5830,0000000000	5190,0000000000	
17,93750000	6598,0000000000	5856,0000000000	
17,95833333	7256,0000000000	6423,0000000000	
17,97916667	7786,0000000000	6876,0000000000	
18,00000000	8187,0000000000	7211,0000000000	

This button opens the **View Data** window.

From the **View Time Data** window, time data can be exported as **XLS (Excel) file**.

Save snapshot of the data window as bitmap image (4)

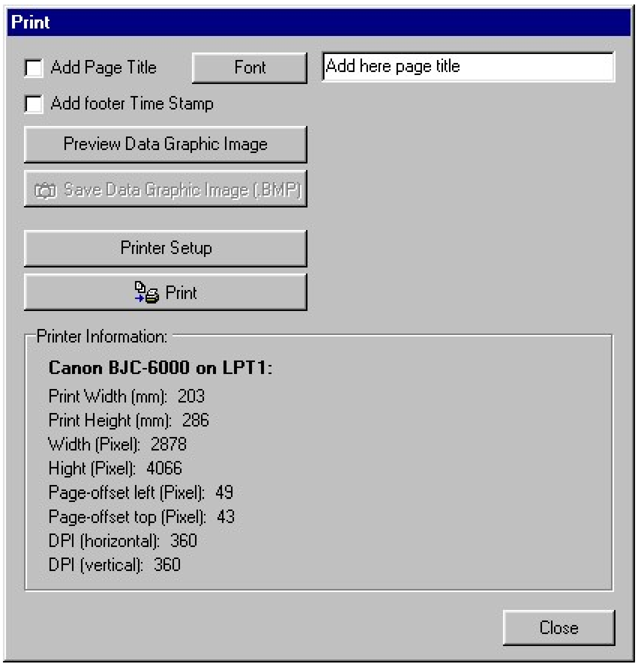
This function allows to save the current plot as a bitmap image (BMP). A dialog is opened when the button is pressed:



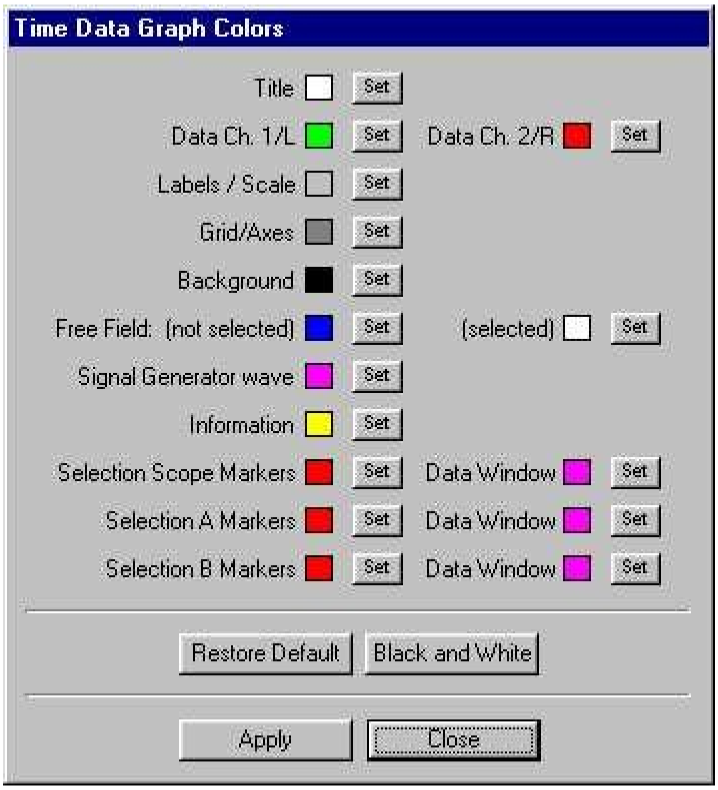
The user can now select the desired resolution for the bitmap image. After this choice, the generated image can be previewed before saving.

Print the data window (5)

This function allows to print the current plot. When the button is pressed, the above **Image Options** dialog is opened. Here the resolution of the printed image can be selected. After this choice, the generated image can be previewed before printing.



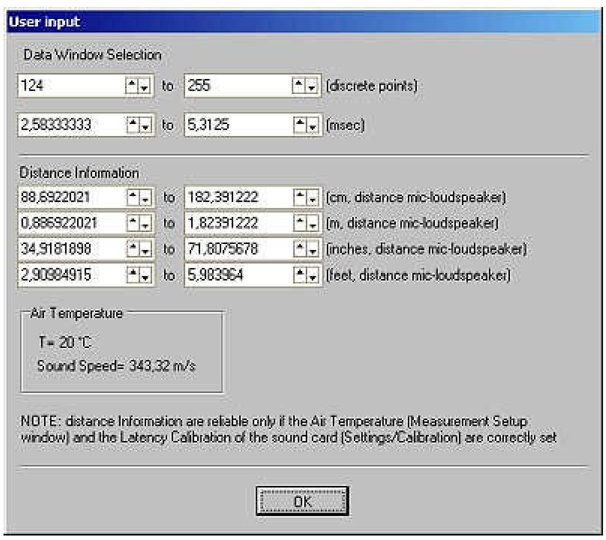
Set custom colors (6)



Here custom colors can be assigned to plota. A **Black and White** preset is available.

Data Selection (for frequency analysis) and Quick Spectrum (7) (8) (9) (10) (11) (12)

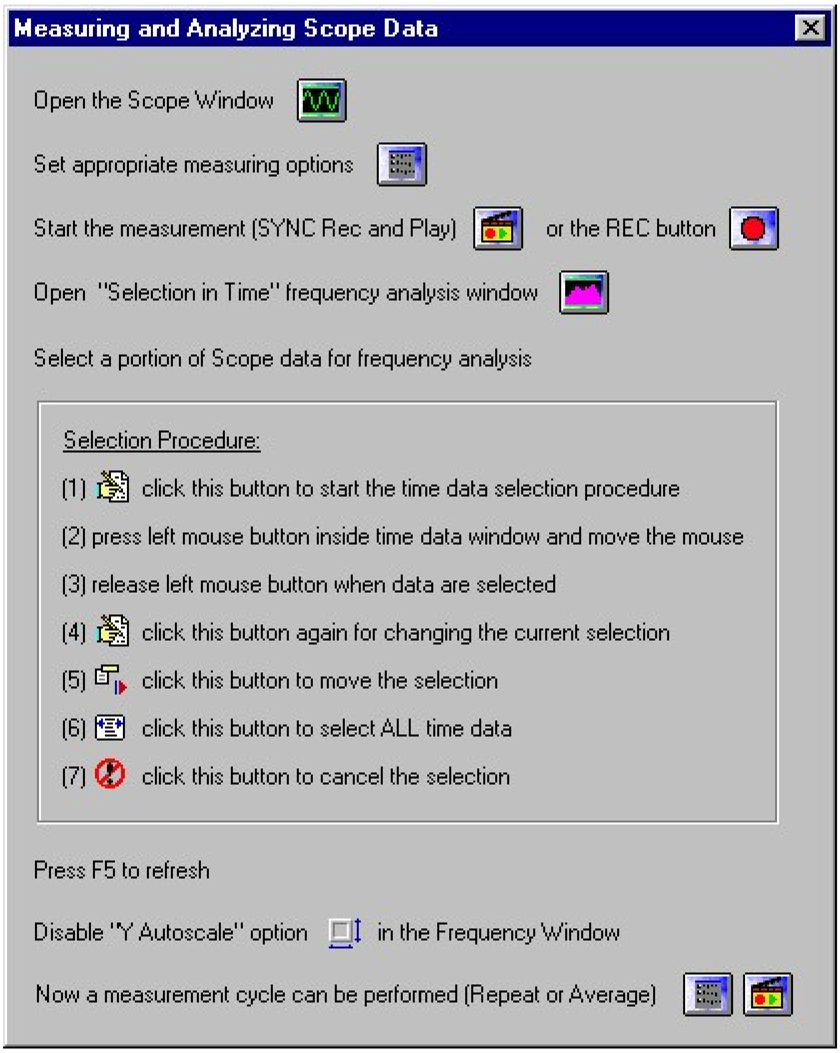
Button (7) can be used for a manual selection of data to be analyzed in Scope window.



Inside the manual selection window other useful information can be obtained:
the microphone-loudspeaker distance can be easily computed from the current data window, just placing the beginning of data window at t=0 and the end of data window at the peak position.
Other distance computations can be done moving the start and end of the data window. This can help to find and verify the true anechoical part of an Impulse Response and isolate the first floor or walls reflections.

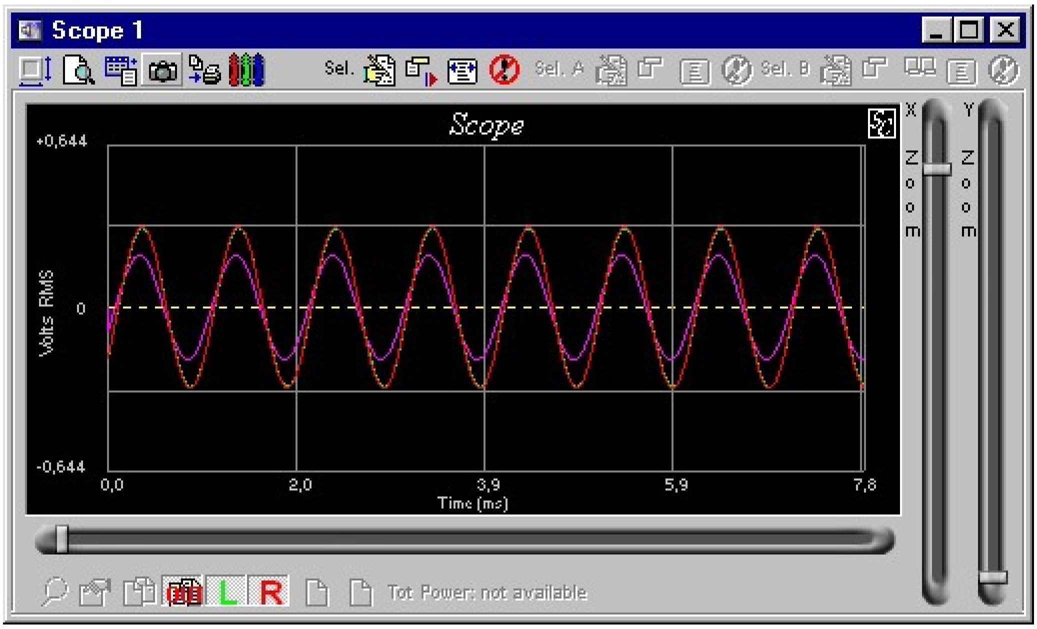
Buttons **(8) (9) (10)** and **(11)** can be used for freely selecting and analyzing Scope Data by means of the mouse.

Click the menu **Help/Quick Help** and select the **Measuring and Analyzing Scope Data** button to see a step by step guide for analyzing Scope Data:



After the data windowing selection, the shortcut **(12) Quick Spectrum** ( ) can be pressed to compute and plot the FFT of the selected data.

● **Overlay generated Signal (14)**



Using this option, the WAVE generated by the internal signal generator can be overlaid with the sampled data (in Scope Mode). This can be extremely useful for comparing the result of a measurement with the Ideal wave in the Time Domain. The example above shows a 200 Hz Square wave (the sampled wave is green and the generated wave is fuchsia).

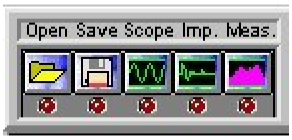
● **PLOT Left and/or Right channel data (for stereo measures) (15)**

This function can be used for plotting only one channel, when the measurement is stereo.

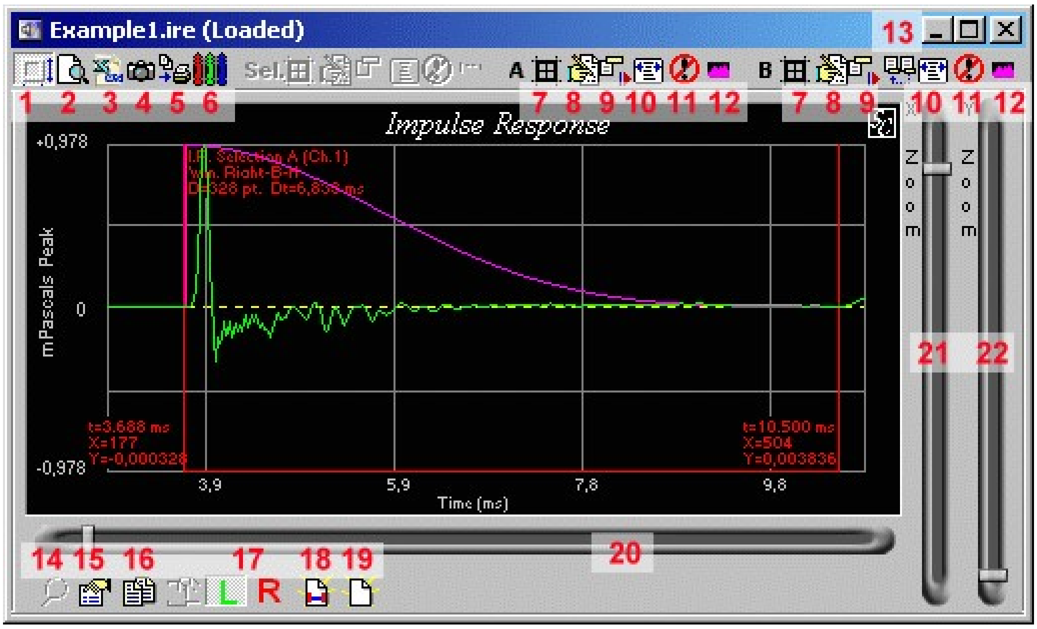
● **Plot Zoom (13) (16) (17)**

Sliders **(13) (16) (17)** can be used for zooming X e Y axis of the plot.
Y Zoom **(17)** is very useful to check negligible signals.

Impulse Response Window



 Impulse Response



1 - AutoScale (ON/OFF)
2 - Set View Time Interval (Lock/Unlock)
3 - View data values / Export time data as XLS (Excel) file
4 - Save snapshot of the data window as bitmap image
5 - Print the data window
6 - Set custom colors

Two impulse response selections (for frequency analysis) are available.
Data Selection "A" and **Data Selection "B"**. Two series of buttons, one for Selection “A” and one for Selection “B” are available, with same functions.

7 - Manual Data windowing and additional information
8 - START selection (if no data are selected) or CHANGE selection size (if some data are selected), moving the selection cursor nearest to the mouse pointer
9 - MOVE selection
10 - EXTEND selection to the current maximum input length or maximum FFT size
11 - DISABLE selection
12 - Quick Spectrum
13 - CREATE selection B with the same size as selection A, starting at the end of selection A

14 - MOVE the view point to localize the maximum peak of the impulse response (the button is enabled only if the impulse peak is outside the plot view range)
15 - SHOW extra information about the impulse on the plot area
16 - OVERLAY the Free Field impulse response (if selected)
17 - PLOT Left and/or Right channel data (if the measurement is stereo)
18 - CLEAR (set to 0) impulse response data between *Selection A* cursors
19 - CLEAR (set to 0) ALL impulse response data
20 - SET plot starting point
21 - ZOOM time scale
22 - ZOOM amplitude (this affects only the plot)

- Pressing the impulse response button an empty window will open.

- When one or more measurements are performed, the Impulse Responses computed from the sampled data are plotted here at the end of every cycle.

- If the **Average** option is enabled, the plot is updated at every measurement cycle with the partial result of the average operation.

Impulse Response Window Features:

Functions **(1)** to **(6)** are quite similar to those described for **Scope** window.

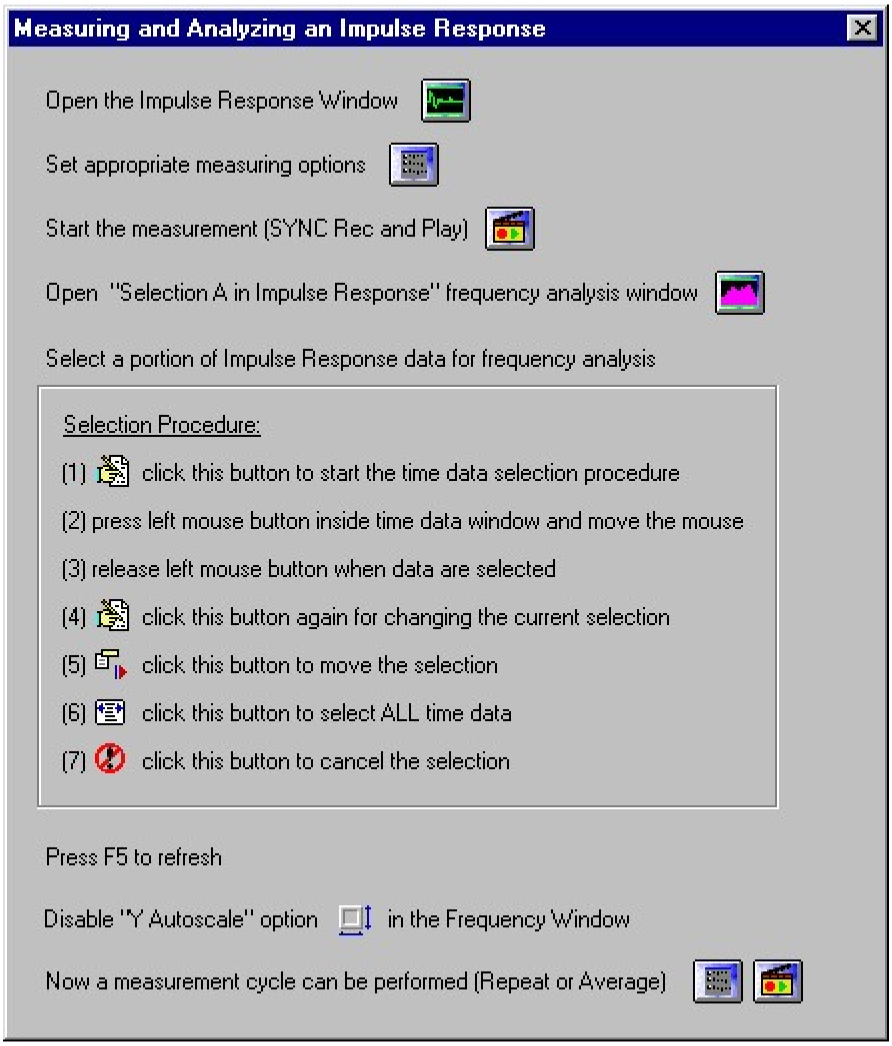
- **Data Selection "A" and "B" (7), (8), (9), (10), (11), (12), (13)**

These buttons can be used for selecting and analyzing Impulse Response Data. Two data selections are available (called **A** and **B**). The frequency contents of the data selected in these selections can be viewed separately and independently.

Buttons **(7)** can be used for manual selection of data to be analyzed in Impulse Response window, as described for Scope window.

Click the menu **Help/Quick Help** to see a step by step guide for analyzing Impulse Response Data.

Inside the **Quick Help window**, by selecting the **Measuring and Analyzing Impulse Response Data** button, the following short Step by Step Guide will be shown:



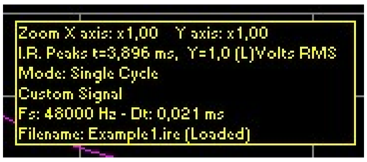
After the data windowing selection, the shortcuts **(12) Quick Spectrum** () can be pressed to compute and plot the FFT of the selected data. A shortcut for Selection A and one for Selection B are available.

Button **(13)** creates selection B with the same size as selection A, starting at the end of selection A.

MOVE the view point to localize the maximum peak (14)

This function can be used for moving the view point (only the view) and localizing quickly the maximum peak of the impulse response. The button (a little lens) is enabled only if the impulse peak is outside the plot view range.

SHOW extra information about the impulse on the plot area (15)



This function shows some extra information about the measured Impulse Response.

OVERLAY the Free Field impulse Response (if selected) (16)

This function allows to overlay two Impulse Responses. A reference I.R. can be set in the Settings/Compensation window. Pressing the button **(16)**, the reference I.R. Is plotted over the current I.R.

Note 1: the sample rate of the reference and current impulses **must** be the same. The reference I.R. color can be set by the user (see custom color function).

Note 2: the I.R. files (**.ire**) store also data selection information. Reference I.R. data selected or unselected can be plotted using different colors (see again custom color function).

PLOT Left and/or Right channel data (if the measurement is stereo) (17)

This function can be used for viewing only one channel, when the measurement is stereo.

CLEAR (set to 0) impulse response data between *Selection A* cursors (18)

This function can be used for zeroing a portion of the data.

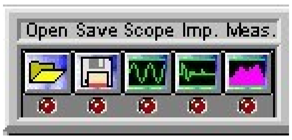
CLEAR (set to 0) ALL impulse response data (19)

This function can be used for setting to 0 all data in the memory bank.

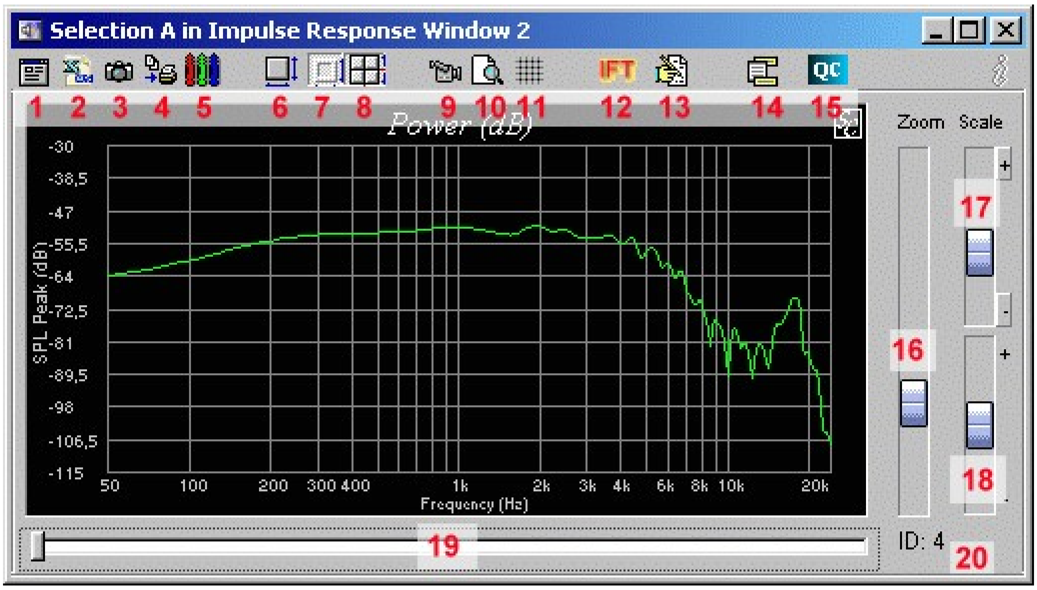
Plot Zoom (20) (21) (22)

Sliders **(20) (21) (22)** can be used for zooming X e Y axis of the plot.
Y Zoom **(22)** is very useful to check negligible signals.

Frequency Domain Window



FFT of Selection (Frequency Analysis of Scope Data),
FFT of Selection A (Frequency Analysis of Impulse Response Data)
FFT of Selection B (Frequency Analysis of Impulse Response Data)



1 - View Options
2 - View frequency data values (ON/OFF)
3 - Save snapshot of the data window as bitmap image
4 - Print the data window
5 - Set custom colors

6 - Y Autoscale on current data
7 - Y Autoscale (ON/OFF)
8 - Round dB Y scale (ON/OFF).

9 - X Zoom out full (display all the measurement frequency band)
10 - Set view frequency range (ON/OFF)
11 - Set logarithmic Y range (ON/OFF)

12 - IFT. Inverse Fourier Transform

13 - Edit Frequency Data.

14 - Frequency Data Overlay

15 - Quality Control.

16 - Frequency zoom. This slider changes the maximum frequency value on the plot
17 - Set the amplitude zoom plotting value
18 - Set bottom plotting value
19 - Set the starting frequency value in the plot

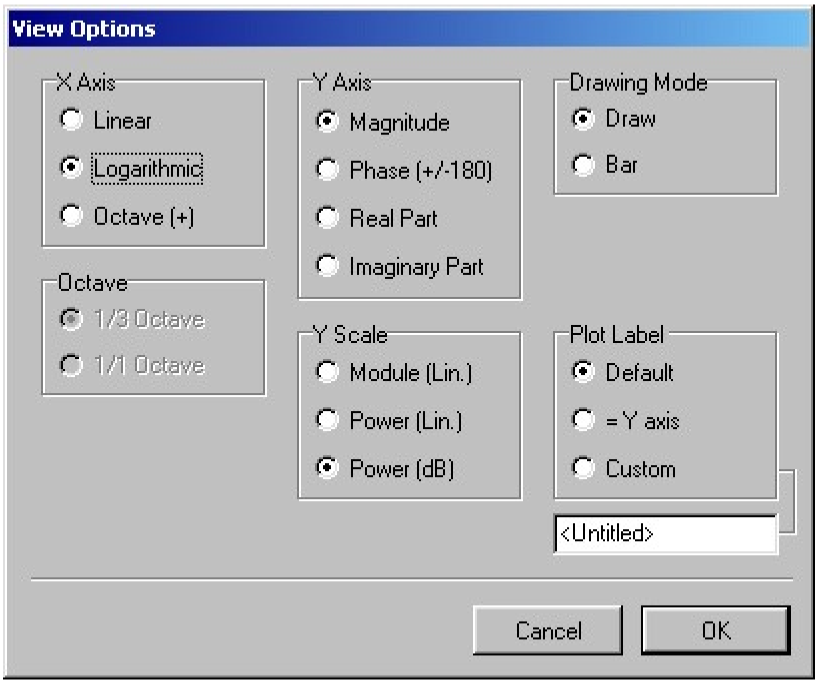
20 - Show frequency data types (ID):

This window shows frequency domain data computed from a selection in the scope window or in the Impulse Response window. It can display also the results of a computation performed by a plugin working in the frequency domain.

Frequency Domain Window Features:

View Options (1)
This function opens the View Options dialog in order to select the view mode. The values shown on the Y scale depend also on the calibration settings (see **Settings/Calibration Window**).

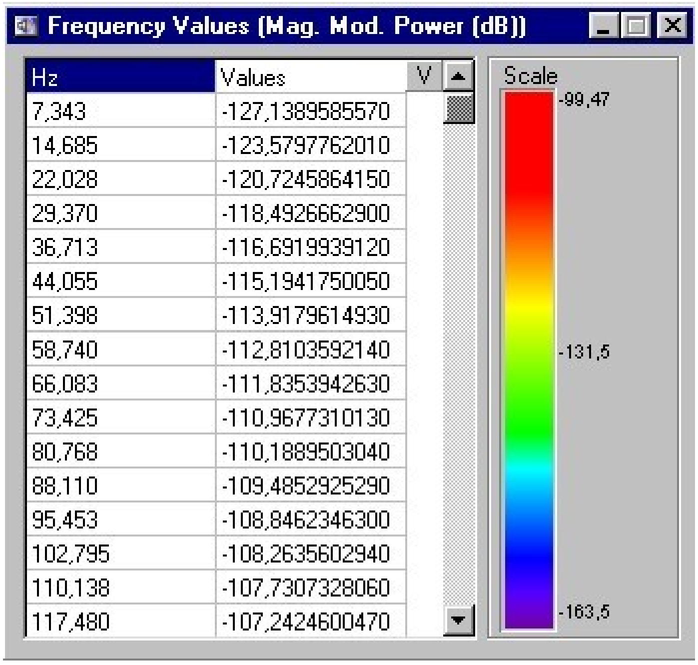
The **drawing mode** option specifies whether the data are plotted as points connected with a line or every data value is plotted as an horizontal bar. The **Bar mode** is much faster and should be used when the CPU is overloaded.



The label of frequency plots can be customized,.allowing to the user the personalization of the title label of the frequency plot, adapting it to the kind of data analyzed and to current settings.

View frequency data values (ON/OFF) (2)

This window shows the values of the currently plotted data. The frequency step depends obviously on the current FFT size (see **Settings/FFT Window**).



From the **View Frequency Data** window, data can be exported as **XLS (Excel) file**.

Save snapshot of the data window as bitmap image (3)

This function allows to save the current plot as a bitmap image (BMP). The user can select the desired resolution for the saved image.

Print the data window (4)

This function allows to print the current plot. The resolution of the printed image can be selected.

Set custom colors (5)

Custom colors can be assigned to the frequency domain plot. A Black and White preset is available.

Y Autoscale on current data (6)

When this button is pressed (once), the minimum and maximum Y plot values are set in an automatic way.

Y Autoscale (ON/OFF) (7)

When this function is ON (the button remains down) the autoscale on the frequency plot is computed at every measurement cycle.
Tip: This function can be disabled after the first measurement. If some measured data are out of the plot, the **Y Autoscale on current data (6)** button can be pressed once.

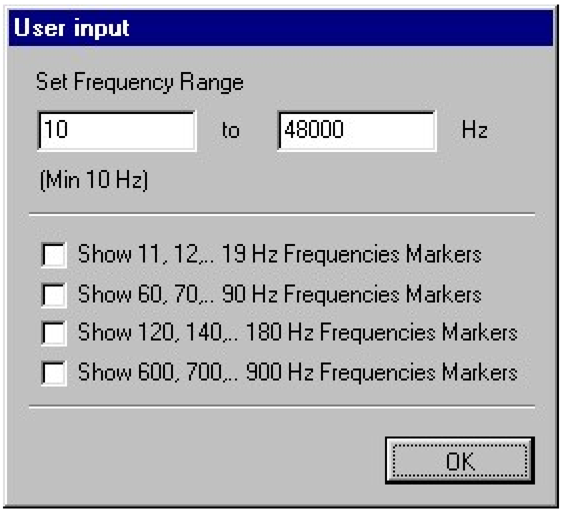
Round dB Y scale (ON/OFF) (8)

When this function is ON and the Y is in dB scale, only integer values are shown. This applies only to the plot, data are untouched

X Zoom out full (9)

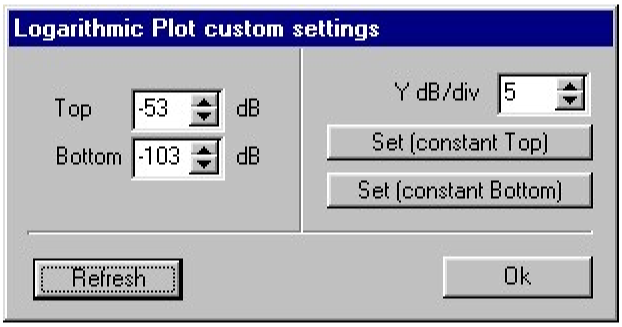
When this button is pressed, all the measurement frequencies are plotted.

Set view frequency range (ON/OFF) (10)



This function allows to set accurately the graph frequency range. When it is enabled, the scroll bars in the Frequency window are disabled. Optional frequency markers in the low frequency range can be activated by the user.

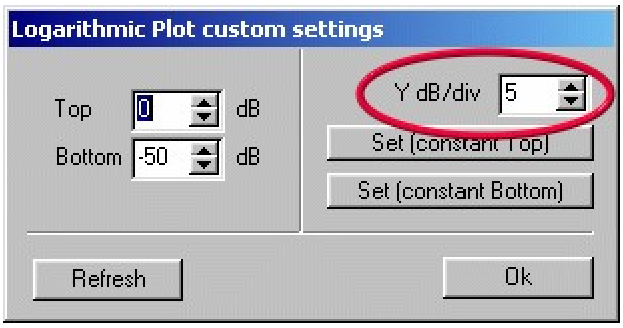
● **Set logarithmic Y range (ON/OFF)** **(11)**

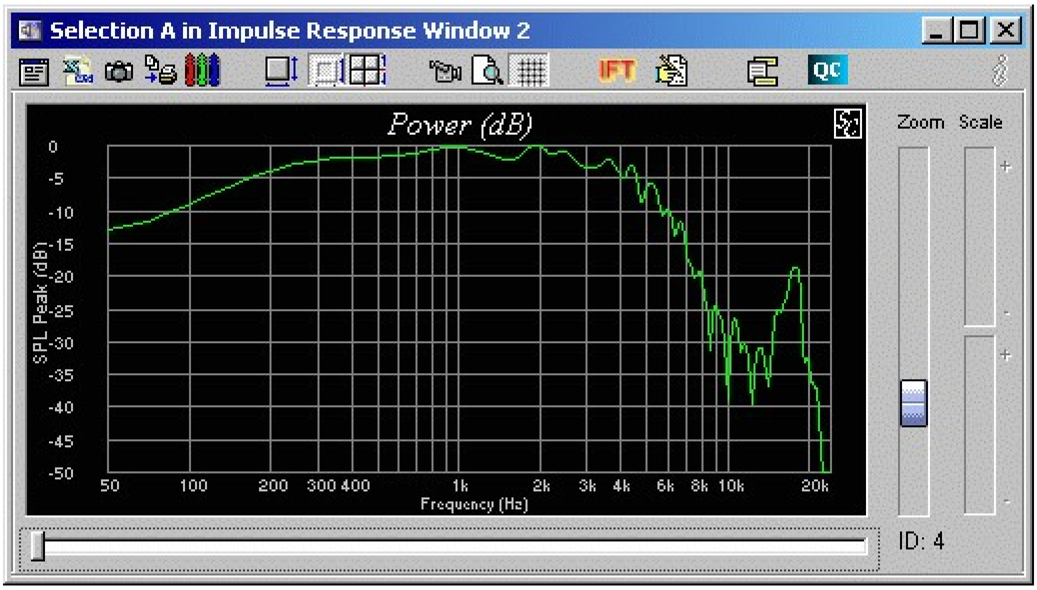


This function allows to set accurately the graph frequency range. When it is enabled, the scroll bars in the Frequency window are disabled. Some options about frequency markers can be set too.

Also a custom dB/div value can be set. Example:
- set the desired Top Y value (in dB)
- set the desired dB/div value (in dB) and press the **Set (constant Top)** button
- press **Refresh**

Now the frequency plot is updated with the set Y scale values.





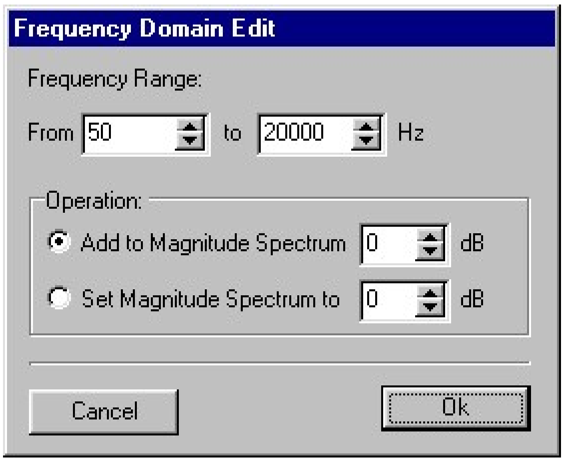
IFT. Inverse Fourier Transform (12)

This function performs the Inverse Fourier Transform of the spectral data content (when possible). Depending on the type of spectrum (see ID type) the resulting time data are stored as scope data or impulse response data.

Sample Champion saves and load spectra as .SPE files, including Real and Imaginary components, in order to achieve a perfect reconstruction of the time data by means of the IFT.

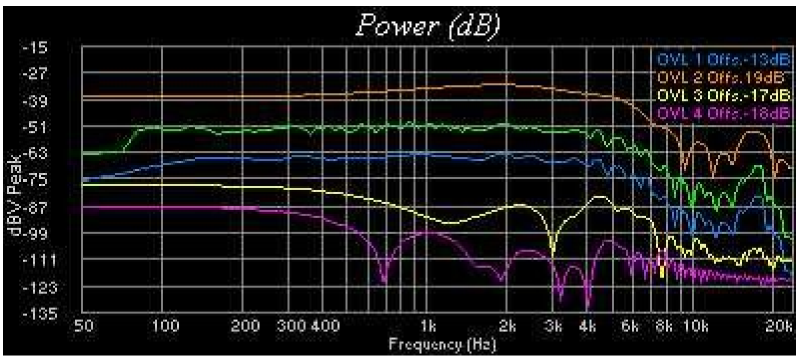
The IFT button also automatically opens a time window containing the result of the Inverse Fourier Transform

Edit Frequency Data (13)



This feature implements a simple frequency edit function.

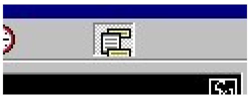
Frequency Data Overlay (14)



Frequency data (in any view mode) can be overlayed. Up to **5 curves** can be simultaneously displayed (current data and 4 overlays). Frequency data drawing mode must be set to "Draw" (not "Bar"), Y Axis and Y Scale can be freely selected. The X Axis

can be Linear or Logarithmic.

The overlay graphs can be set and enabled by clicking the **Overlay** button in the frequency data window.



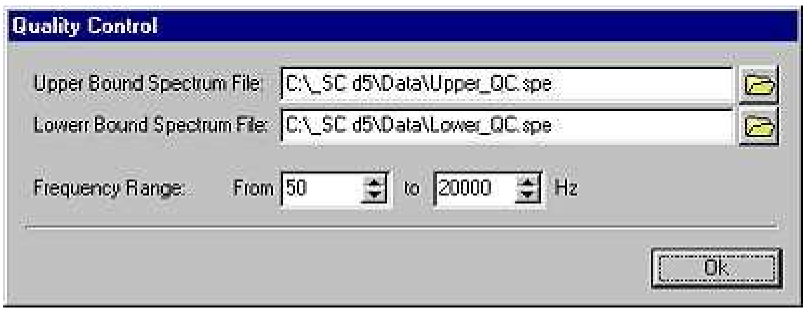
This will open the Overlay Frequency Data window.

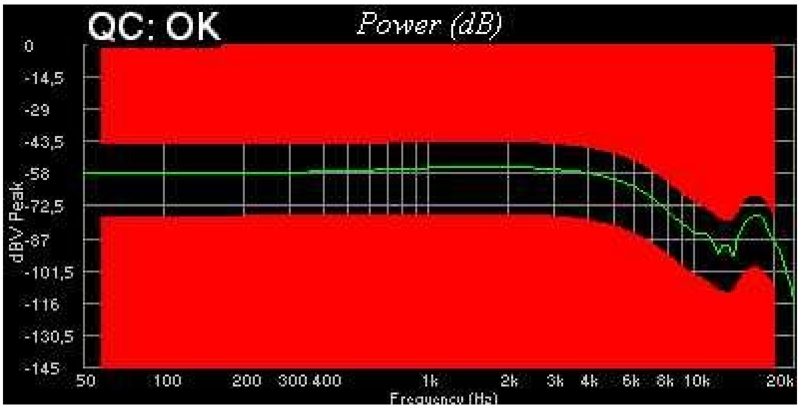
By clicking the "Set" button, the currently plotted spectrum will be set as overlay curve (and saved in the default data directory with the name shown, as standard ***.SPE*** file). Also all previously saved .SPE files can be loaded for overlay, provided that they have the same FFT size and sampling frequency of currently displayed spectra. The colors of the overlay curves can be changed by clicking the "Color" buttons. The settings of the overlay curves are saved when the Overlay window is closed (by clicking again the Overlay button).

A relative **offset** in Frequency Overlay can now be selected by the user (adding or subtracting the desired amount in dB). Spectra measured with different amplitudes can be thus easily compared.

Quality Control (15)

This feature allows to assign upper and lower spectra. The computed spectrum is then compared, for each frequency, with the upper and lower limits. If even a single value is outside bounds, a **BAD** message is written, otherwise the result is **OK**.





Load **Example_QC.spe** and set **Upper_QC.spe** and **Lower_QC.spe** (included in the setup package, Data directory) to see an example.

Show frequency data types (ID) (20)

2 = FFT of a selection from scope data
4 = FFT of selection **A** from IR data
6 = FFT of selection **B** from IR data
other = frequency data from plugins (each plugin has his own ID type)

General notes about frequency data computation

The computed frequency data depend on to current FFT settings (FFT size, weighting window, cursor selection, phase shift and transfer function measurement).
See **Settings / FFT**.

The **phase shift** option is always enabled for **averaged** FFT of **scope** data.

WARNING: the spectral data are recomputed EVERY TIME that the refresh button (or the F5 key) is pressed. In the following cases **special attention** must be paid:

(1) loading spectral data of the same type of a currently plotted computed data

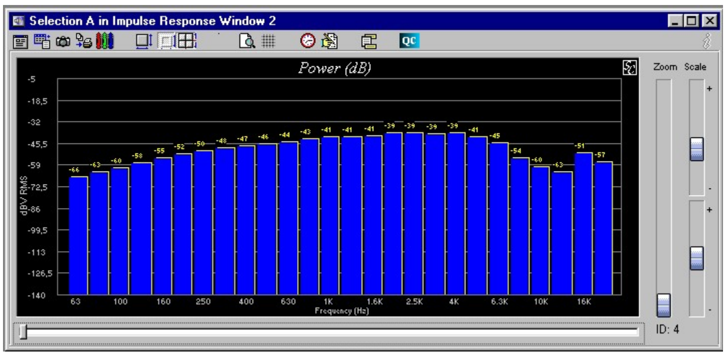
(2) analyzing frequency domain data from a scope selection after an average measurement cycle

(1) For instance, **if**

- the **impulse response** window is open
and
- selection **A** on the impulse data is active
and
- the **frequency domain window** of selection **A** is open
then
WHEN A FILE CONTAINING FREQUENCY DATA FROM SELECTION **A** IS OPENED, THE LOADED DATA WILL REPLACE THE COMPUTED DATA **UNTIL A REFRESH IS PERFORMED**.
This is true for every file type.

(2) For instance, **if**
- the **scope** window is open
and
- selection on the scope data is active
and
- the average mode is active and a measure cycle has been performed (number of blocks greater than 1)
and
- the **frequency domain window** of the scope selection is open
then
THE FREQUENCY DATA PLOTTED WHEN THE MEASUREMENT CYCLE IS COMPLETED ARE THE DATA AVERAGED (IN THE FREQUENCY DOMAIN) **UNTIL A REFRESH IS PERFORMED**. WHEN A REFRESH IS PERFORMED AND THE SELECTION IS ACTIVE, THE PLOTTED DATA WILL BE REPLACED WITH FREQUENCY DATA COMPUTED FROM THE LAST SAMPLED SCOPE DATA (THE CURRENTLY PLOTTED DATA).

When the Octave mode is selected, the value of each band is shown above the band if enough space is available (this can be obtained by enlarging or maximizing the window).



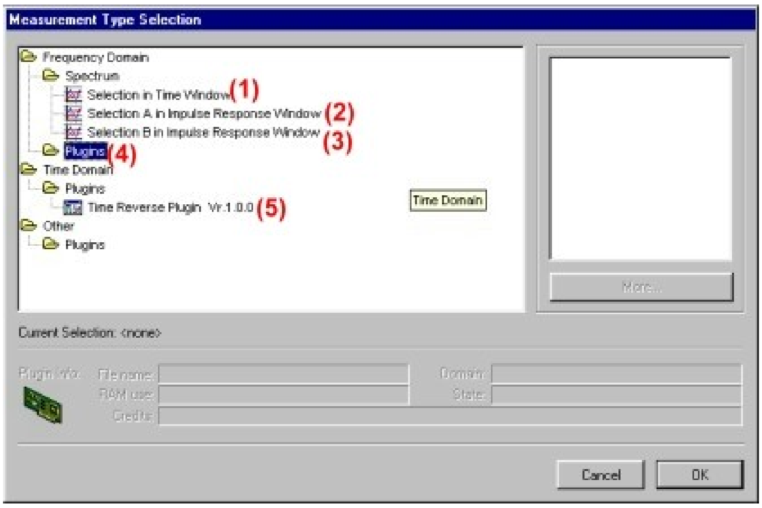
See also **Post-Processing menu**.

Measurement Type Selection

Measurement type selection

- Pressing this button the measurement type selection window will open. Here it is possible to select which type of analysis/measurement must be performed on the sampled data. This dialog can be used for opening the **Frequency Domain Window** above.

- In this window it is also possible to open or close plugins.



- Three standard types of analysis are available:

  (1) Selection in Time Window
  (2) Selection A in Impulse Response Window
  (3) Selection B in Impulse Response Window

- **Selection in Time Window (1)**

Clicking here, the program will open a window showing **frequency domain data** (magnitude, phase, real or imaginary part) of the **scope window data selection**, using **current FFT settings**.

FFT selections can be opened / closed also by means of shortcuts buttons on the **Control Bar** and using the **Quick Spectrum** button ( ) on the Time and Impulse Response windows.

Selection A in Impulse Response Window (2)

Clicking here the program will open a window showing **frequency domain data** (magnitude, phase, real or imaginary part) of the **impulse response window data selection A**, using **current FFT settings**.

Selection B in Impulse Response Window (3)

Clicking here the program will open a window showing **frequency domain data** (magnitude, phase, real or imaginary part) of the **impulse response window data selection B**, using **current FFT settings**.

Frequency domain plugins (4)

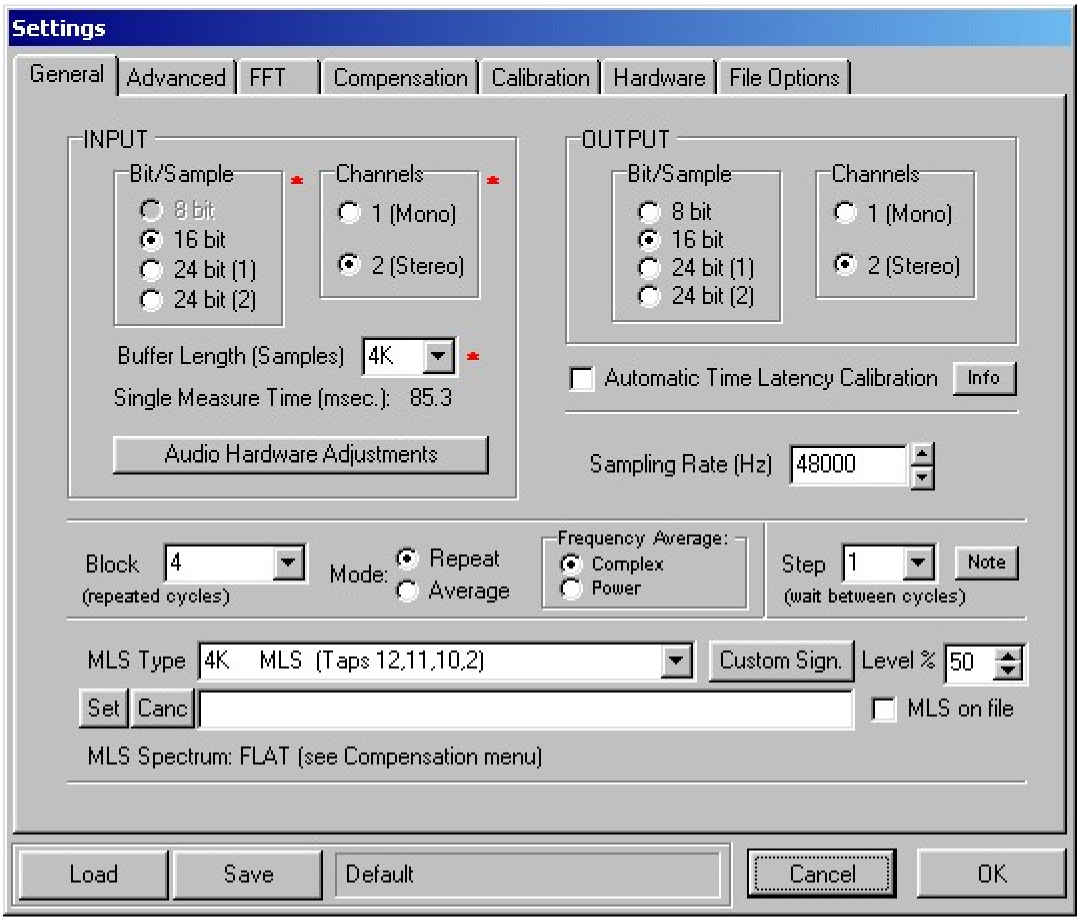
Here the program will show the list of available **frequency** domain plugins. Selecting one of them, the program will open the corresponding plugin. No frequency domain plugins are available in the above picture.

Time domain plugins (5)

Here the program will show the list of available **time** domain plugins. Selecting one of them, the program will open the corresponding plugin. One time domain plugin is available in the picture above.

Settings - General

Settings - General



- The **input** (recording) and **output** (playing) PCM format can be set in this window.

- **24 bit sampling.** Two modes have been implemented:

- 24 bit (1) uses an internal data block align of 3 bytes (actually 24 bit). This mode is suggested for sound cards with A/D and D/A converters with bit resolutions of 18 or 20 bit (for example Pinnacle sound card)

- 24 bit (2) uses an internal data block align of 4 bytes (actually 32 bit). This mode should work with all 24 bit sound cards.

The **Sampling Rate** is the same for IN and OUT. When the settings window is open or the sampling rate is changed, the program performs a silent full-duplex synchronous PLAY/REC test. If the selected sampling frequency is not supported by the sound card, a red alert message is shown. Note that the alert is shown also when the audio card is busy (for example is currently used by another program).

Every sampling rate is supported in post-processing mode, for example loading a TIM file sampled with MLSSA™ system at any sampling rate.

A CUSTOM SAMPLE RATE CAN BE SET JUST BY TYPING IT IN THE FREQUENCY BOX. Using the arrows near the frequency box, the standard sample rates can be selected.

The **Buffer Length** (input buffer) is actually equal to the value selected - 1 (for example 8K means 8*1024-1=8191 samples), in order to have the same length of the corresponding MLS sequence.

The **Block** value indicates how many measurement cycles must be performed before stopping the sampling.
This value can be changed also in the Quick Settings Window.

The **Mode** option indicates whether measured data (if Block>1) are averaged or the data measured at every cycle overwrite the data of the previous measurement. **This setting can be changed also in the Quick Settings Window.**

When the **Power Frequency Average Mode** is selected, the magnitude is averaged and the phase information is ignored (set to 0)
The power average mode can be useful when sampling pure tones generated by an external source (asynchronously). Note that when a spectrum averaged in power mode is saved, the **.SPE** file is compatible with all other .SPE files (a vector of complex data, positive and negative frequencies), but the **Real part** of the data is the magnitude of the spectrum while the **Imaginary part** of the data is **0**.

WARNING: When the **Complex Frequency Average Mode** is selected, the average in the frequency domain is actually computed on complex data. This kind of average should be performed **ONLY** by using the internal signal generator, to assure synchronism and coherence between all measurements.
This setting can be changed also in the Quick Settings Window.

When the audio **input** format parameters are changed, the **output** format will be changed accordingly. Also a Buffer Length change will cause the automatic selection of a right length MLS signal. Output parameters and MLS signal settings can still be changed manually.

NOTE: **Automatic Latency Calibration** option can be used only when 2 Channels INPUT / OUTPUT format is selected. When this option is enabled, the soundcard Channel 2 INPUT must be connected directly to Channel 2 OUTPUT (Loopback).
See **Automatic Latency Calibration** for more information.

There is some difference between **averaging** in time and frequency domain:

- **Time domain**:

  Impulse Response data are averaged in the **TIME DOMAIN**

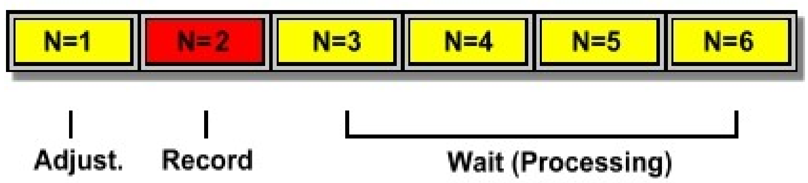
  Scope data are **NEVER** averaged.

- **Frequency domain**:

  FFT from Impulse Response are **NEVER** averaged in the **FREQUENCY DOMAIN** because they are computed from data already averaged in the time domain

  FFT from Scope: are averaged in the **FREQUENCY DOMAIN**

- The **Step** value indicates the number of **wait** cycles **after** the actual record cycle. Wait cycles are necessary to the program for performing all processing operations (recovery of impulse response from MLS, FFT of selected data and other user selected processing). The program performs also a single wait block **before** the actual recording cycle, to adjust some internal settings and allow to the acoustic system to reach a stationary state. This single block is not counted in the **Step** value. The signal generator is active during wait states, in order to leave the acoustic system excited and maintain the synchronism with the sampler.



In the example in the picture above the value of **Step** is equal to 4

If, for example:

Block = 8
Step = 4

then the complete cycle of measures (repeated or averaged) requires 8*(1+1+4) = 48 recording cycles (each one containing **Buffer Length** samples).
This value can be changed also in the Quick Settings Window.

From the **MLS Type** list, it is possible to select the length and type of the MLS stimulus or other signals

Note that when **MLS length** and **Buffer length** are not equal, the program will show a red alert message inside the Impulse Response window.

To achieve the correct impulse response recovery, **MLS LENGTH AND INPUT LENGTH MUST BE THE SAME!**

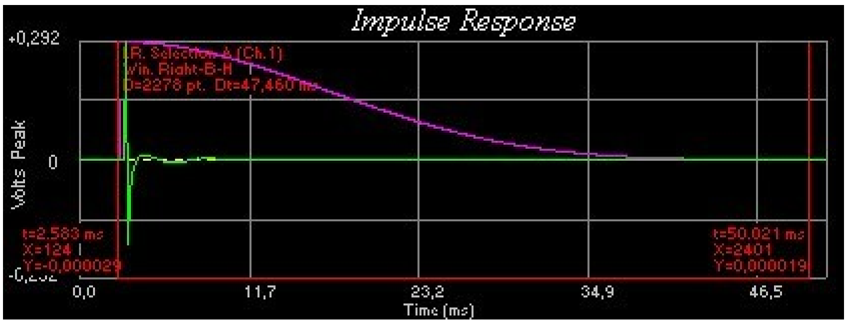
Under the MLS Type menu, the program shows the current MLS spectrum status (see **Settings / Compensation**)

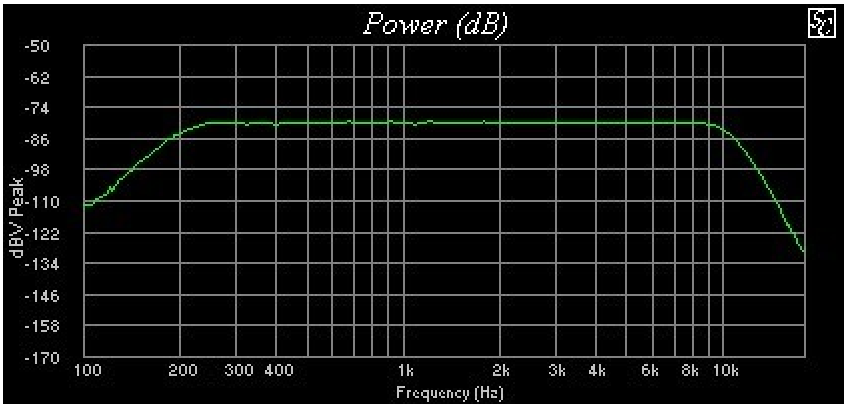
A **custom** WAV file can be used as **MLS** signal. It is possible in this way to pre-filter the MLS signal itself. This could be useful, for example, when performing measurements on tweeters or woofers.

An MLS signal generated by Sample Champion can be saved (Menu File), filtered by means of an external software and set as **custom MLS**.

Tip:
The MLS signal can be filtered by means of the "Filter Banks Plugin" (that works only on Impulse Responses), by using the following procedure:
1- Save the current generated MLS signal (Menu File) as a WAV file
2- Set (in Settings/File Options dialog) "Load WAV file as.. Impulse Response"
3- Load the WAV file (now considered as an Impulse Response)
4- Open the Filter Banks Plugin and filter the I.R. as desired
5- Save the I.R. as a WAV file and set it as custom MLS signal.

Now the filtered MLS can be used. Note that the **MLS type selected** in the Settings **MUST** have the **same order AND TAPS** of the **custom MLS file**, for a proper reconstruction of the Impulse Response.





Example of Loop-Back Impulse Response, measured
using a band-pass filtered MLS signal

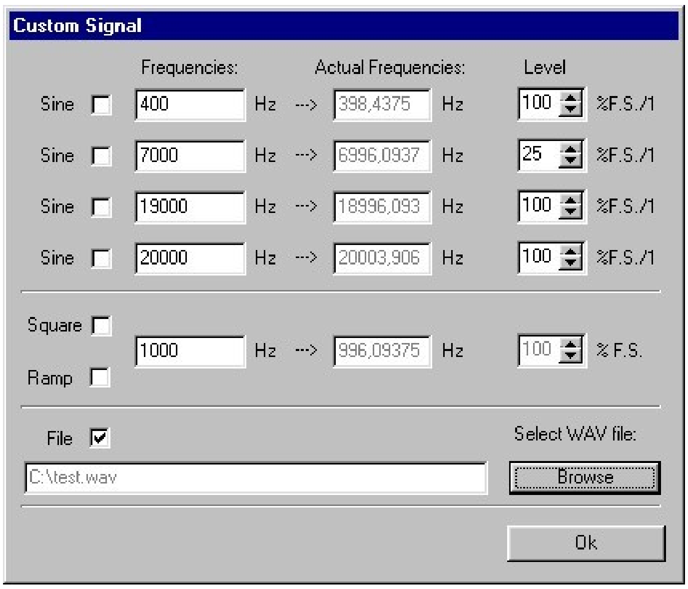
- The **Level** value is the percent level of the generated digital signal, referred to full scale (independently from mixer settings)

- By means of **Load** and **Save** buttons, it is possible to load and save the current global settings from/to an INI file.

Note that the program settings are updated only when the settings window is closed by pressing the OK button.

- The **Custom** signal button will open a window for selecting either a multi-tone, square or ramp signal or a custom WAV file.

The actual frequencies of the generated signals are slightly different from those written by the user, in order to have a perfect correspondence with the discrete frequencies analyzed by the FFT. An increase in the FFT length will reduce the difference between generated and desired frequencies.

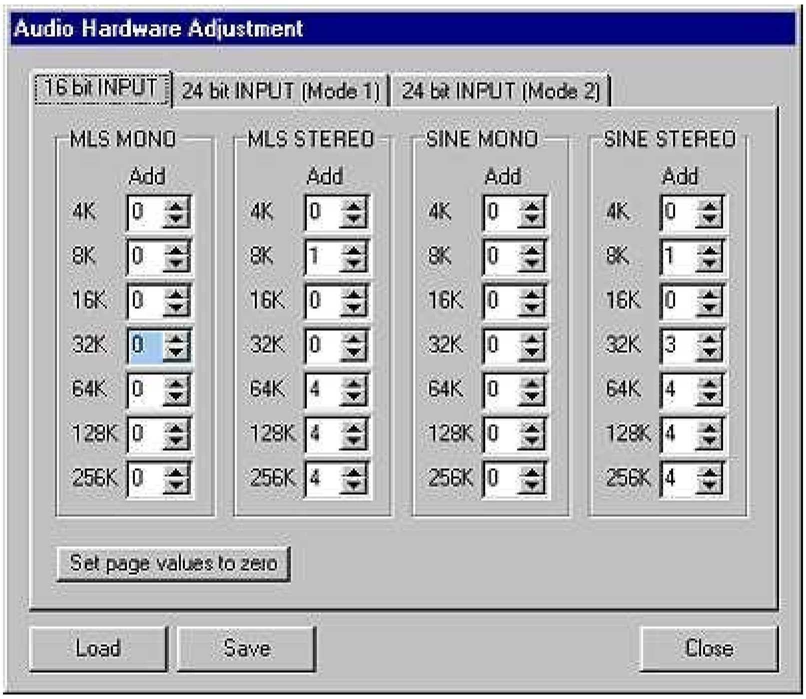


When the **sine wave** signal type is selected, the input and output buffers are set at 2^N samples in order to perform an exact FFT analysis (coherent sampling feature).

When the **square wave** signal type is selected, the generator will play a square signal with the selected frequency **only during one half of the sampling interval**. This is done because the input and output lengths are one sample shorter than 2^N samples and it is not possible to generate a square wave of this length with the right harmonic content. But this is not a problem when performing a measure with this signal: just select an input buffer length 2 or 4 times longer than the FFT size. See Apllication note #5.

- The **Audio Hardware Adjustements** button allows a "fine tuning" of the Audio Card.

Use this function **ONLY** if, in repeated sequences of measurements, the obtained impulse is moving.



Connect the **input** to the **output** of the soundcard with the **loop-back** cable as described in the configuration tips web page and perform a repeated sequence of measurements, opening scope and impulse response windows.

Now different behaviors can be observed:

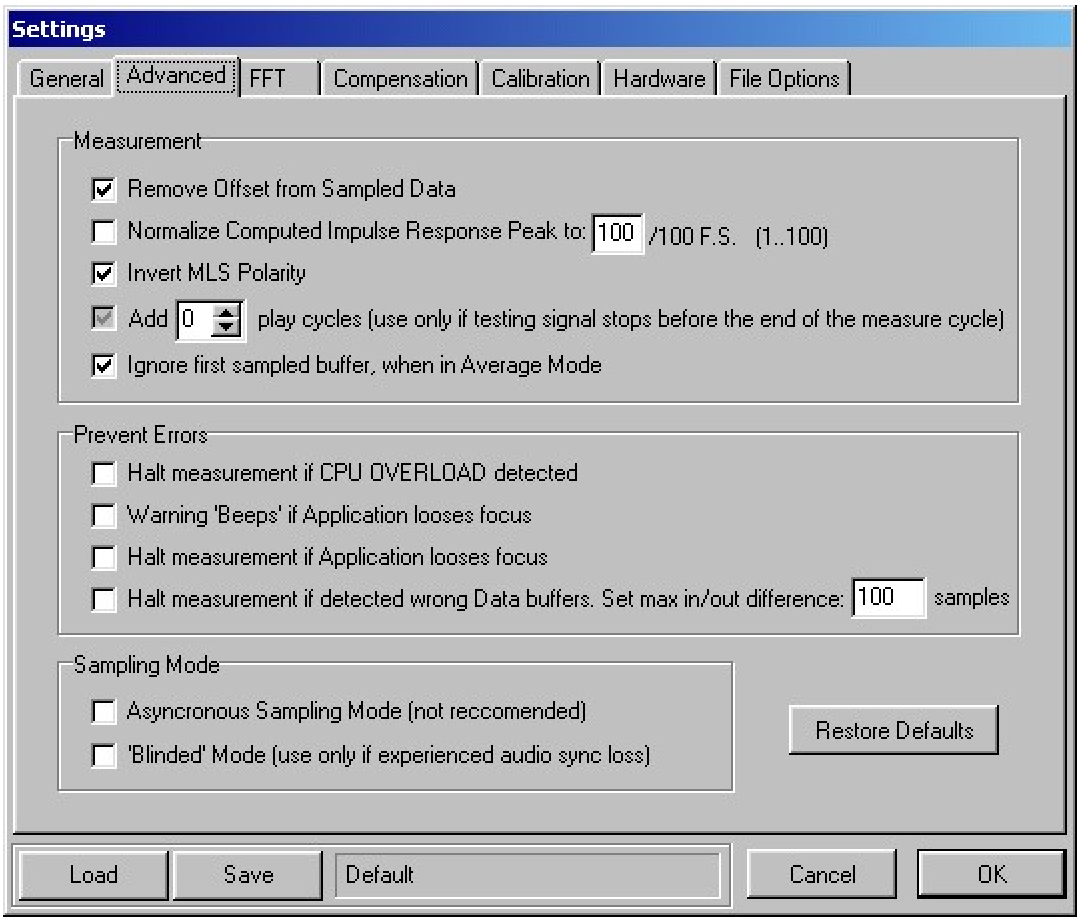
- the **impulse response** could be **still** (in time) and **clean**: in this case the adjustment for this buffer length, channel setting and bit depth does **not require any adjustment** (leave "0"). Check the cleanness by zooming the impulse amplitude to max by means of the Y zoom slider.

- the **impulse** is **moving** and probably is also **not clean** (while in the loop-back measure it should, if input and output levels are correctly set): **increase** the length of this buffer (the current setting is marked in blue in the adjustment window) if the impulse **moves to the right** at every measurement cycle and **decrease** its length if the impulse **moves to the left**. The same adjustment should be done also when the impulse is still but not clean. Check also ALL scope data to verify whether if something is wrong.

NOTE: This function should be used only as last resource, to compensate buggy soundcard drivers!

Settings - Advanced

 Settings - Advanced



Some advanced options can be selected in this window.

MEASUREMENT OPTIONS:

- The **Remove Offset from Sampled Data** option causes the subtraction of the offset present in scope data, **only** for the computation of the Impulse Response. The scope data and FFT from scope data are not affected by this option.

- The **Normalize** option causes the normalization of the calculated Impulse Response peak between 0 and 1, overriding calibration settings.

- The **Invert MLS polarity** option changes the polarity of the stimulus.

- The **Add [number] play cycles** option can be used if the testing signal stops before the end of measure cycle, due to some bugs in the soundcard drivers.

- When the **Ignore first sampled buffer (in Average Mode)** option is enabled the first sampled buffer will not be averaged. This can be useful for compensating some soundcards behavior or for specific measurements requirements.

ERROR PREVENTION OPTIONS:

- **Halt measurement if CPU Overload** option stops the program when the CPU load is excessive.

If CPU Overload is detected, try to:

- increase "Step" value in "General Settings" window

If the error remains, try to:
- increase Buffer length
- decrease FFT size (if FFT is used)
- decrease Sampling Rate
- avoid changing Selections and other controls inside Data windows during measures

- **Warnings beeps** and **Halt measurement if application loses focus** options prevent the user switching to applications that could break or corrupt the normal sampling data flow.

- **Halt measurement if wrong data buffer is detected** option stops the measurement

when the difference (in samples) between IN and OUT buffers is greater than the indicated value. This option works only if the audio card has a very low latency time and must be used carefully. Should the use of this option cause some false alarms, don't worry and disable it.

SAMPLING MODE OPTIONS:

- **Asynchronous sampling mode:** this mode should not be used because all advantages deriving from synchronism between IN and OUT are lost. When activated, the sampling engine restarts at every new measurement cycle. The **average mode** is disabled when this option is selected.

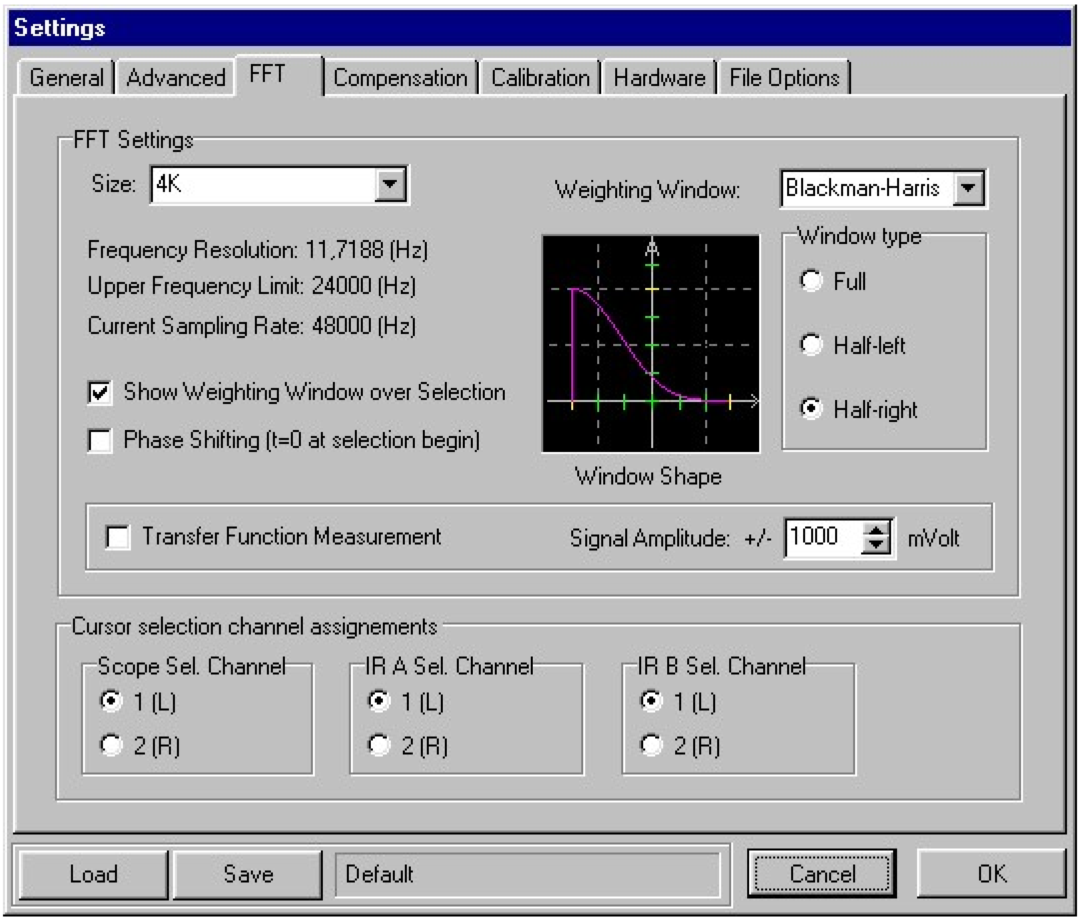
⚠ Asynchronous sampling mode should be used **ONLY** when loss of synchronism is experienced. There is a loss of synchronism when the impulse response peak, in **repeat mode**, is not located at the same time position, in every cycle (see application notes).

- **Blinded' Mode** has been created to prevent any corruption of the sampled data caused by the influence of the graphic plotting, especially on slow computers endowed with old video cards. When this option is enabled, during sampling all graphic windows are hidden and only a small text window is open.

⚠ Blinded mode should be used **ONLY** when a loss of synchronism is experienced. There is a loss of synchronism when the impulse response peak, in **repeat mode**, is not located at the same time position in every cycle.

Settings - FFT

 Settings - FFT



The **FFT Size** value is the number of points used by the current Fast Fourier Transform algorithm. If the selection length on time data is shorter than the FFT size, the missing points are set to zero (zero padding). Note that the implemented algorithm works on double precision complex data and both positive and negative frequencies are computed.

The **Weighting Window** can be chosen among Rectangular, Bartlett, Blackman-Harris, Hanning, Hamming, Blackman shapes. Full, half-right, half-left window types are available. **This setting can be changed also in the Quick Settings Window.**

When **Show Weighting Window over Selection** is checked, the shape of the window is superimposed to the plotted data in the Scope Window or in the Impulse Response Window.

The **Phase Shifting** option, when active, performs a phase shifting on the FFT data. An IFT (Inverse Fourier Transform) on shifted data forces the computed time series to start at *t=0.*

The **Phase Shifting** option is always enabled for **averaged** FFT of **scope** data.

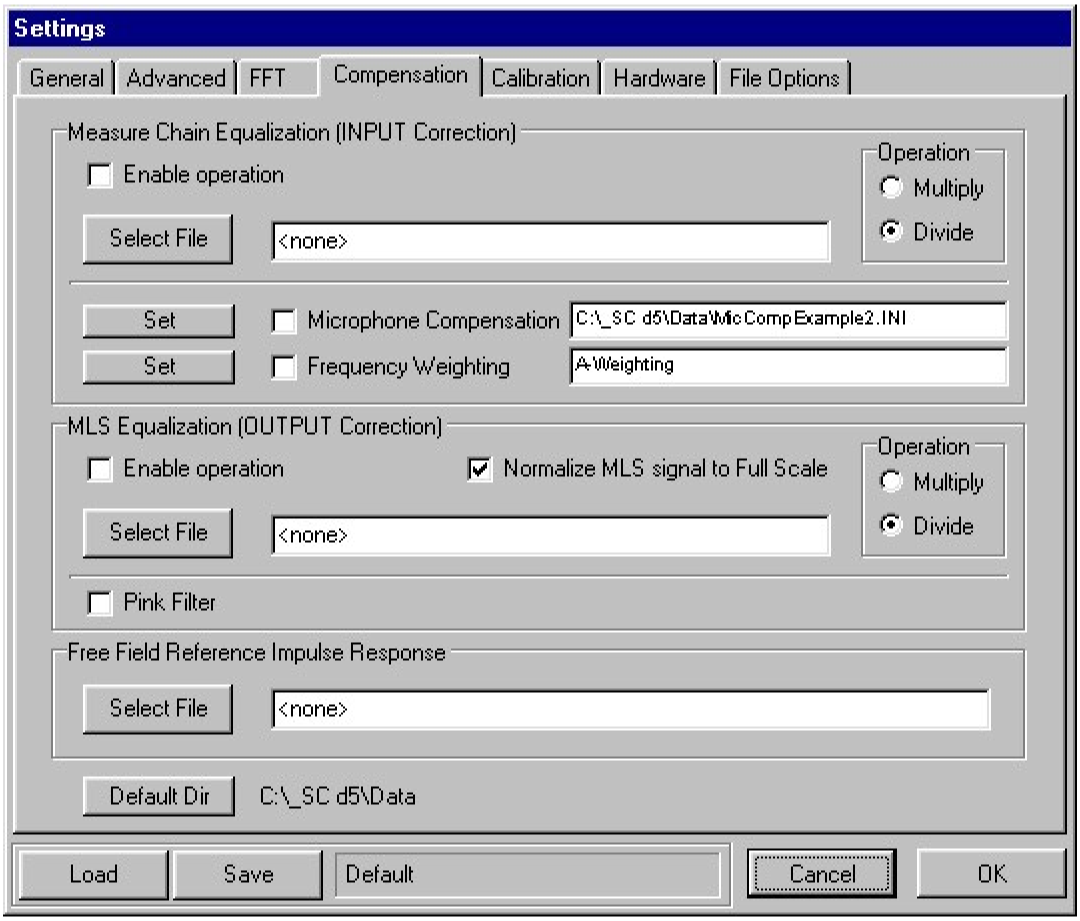
When the **Transfer Function Measurement** option is active, the computed FFT data are normalized by the **Signal Amplitude** voltage value. Use this option when analyzing Impulse Response data.

The **Signal Amplitude** voltage value is the RMS level of the output signal. It must be manually set. Note that this value is expressed in milliVolts.

The **Cursor Selection channel assignments** option assigns one of the two data channels to the FFT, in case of 2 channel measurements.
This function is available also in the Quick Settings Window.

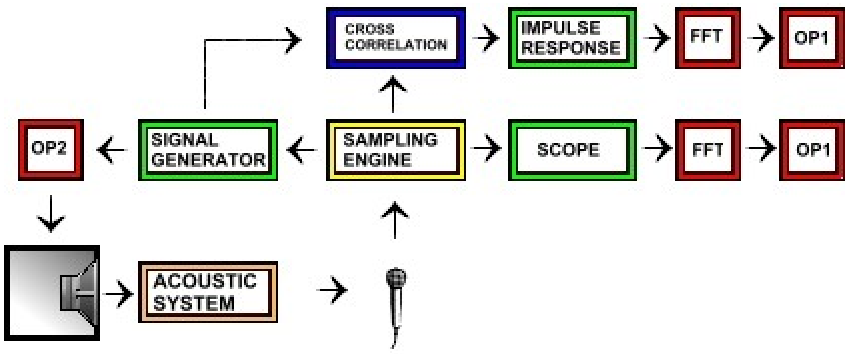
Settings - Compensation

 Settings - Compensation



Here the user can select some pre and post-filtering operations on the sampled data or on the generated signals.

The following flow chart describes the input and output signal paths:



In the chart, *Acoustic System* denotes the system under test, for example a room or a loudspeaker.

When the **Measure Chain Equalization (INPUT Correction)** option is enabled, the content of all frequency domain banks (FFT of a selection of scope or impulse response data) are processed by performing the selected operation (**Multiply** or **Divide**) by the selected frequency domain file (**.SPE**). This operation, represented by **OP1** in the flow chart, is performed on complex numbers. Note that both internal frequency domain banks and .SPE files contain complex numbers and both positive and negative frequencies.

The number of samples of the selected frequency **.SPE** file and the internal current **FFT size** must be the same. Also the **Sampling Frequency** of the selected file and the current **internal Sampling Frequency** must be equal.

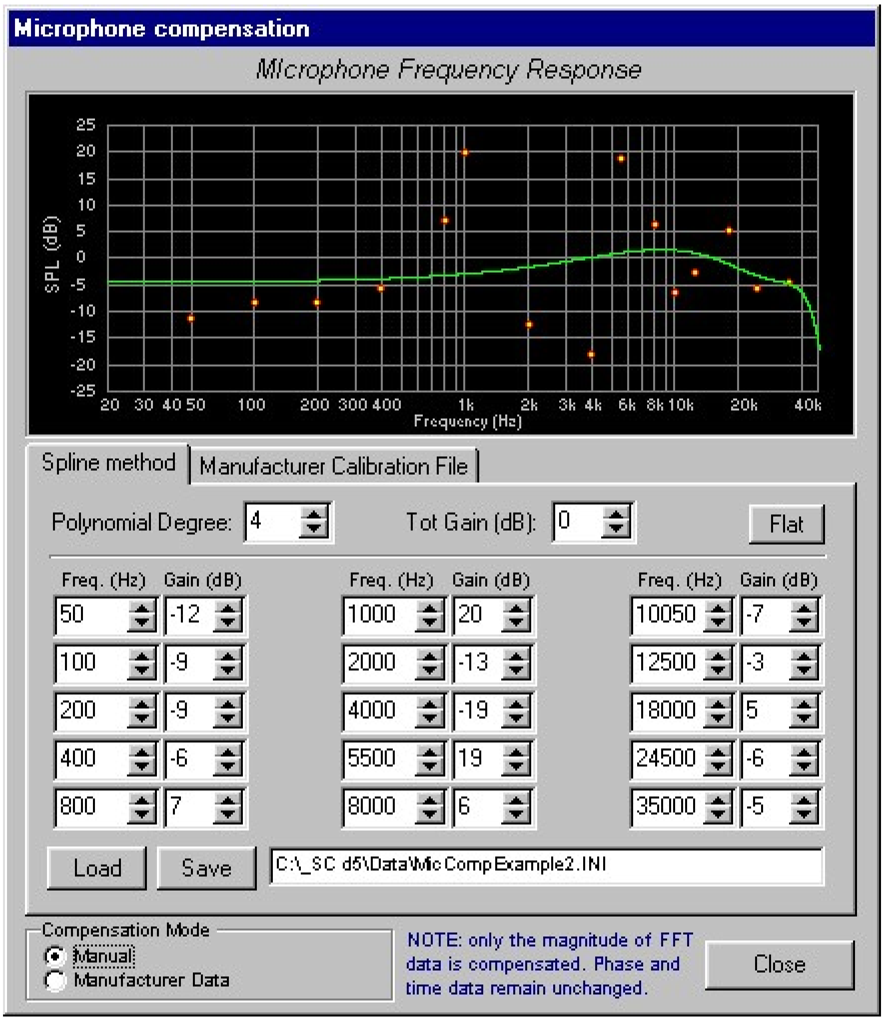
When the **Microphone Compensation** option is enabled, the content of all **frequency domain banks** (FFT of a selection of scope or impulse response data) are processed by performing the compensation that has been selected.

NOTE: only **frequency domain banks** are automatically compensated! Time domain data (Impulse Response) can be manually compensated (see **Post-Processing functions**).

Two methods for reconstructing the **Microphone Frequency Response** are implemented:

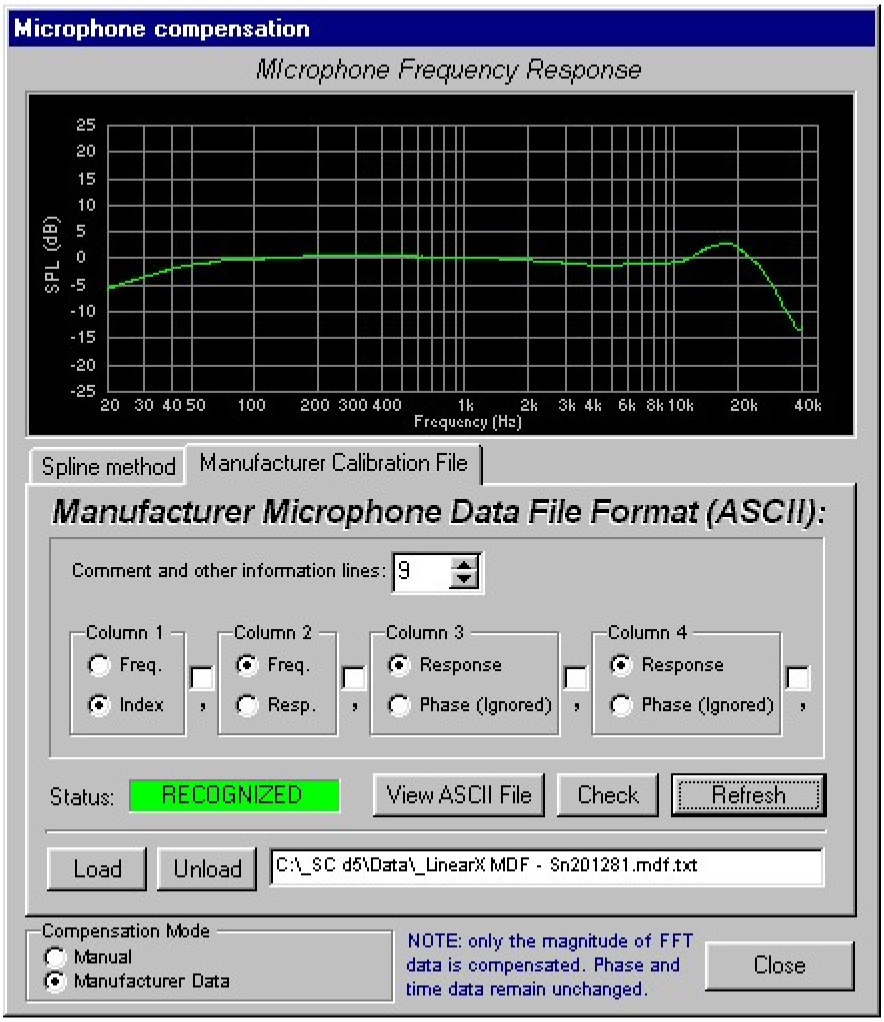
- Spline

The user can manually try to reproduce the microphone frequency respone curve (when only a graph is furnished by the microphone manufacturer). This is an empirical method.



- Frequency Response furnished by the microphone manufacturer.

This function adds the capability of recognizing microphone frequency responses (ASCII format). Now the microphone can be perfectly compensated by means of data furnished by the manufacturer. The range of the microphone compensation extends up to 48 kHz. The format of the compensation ASCII file must be selected manually



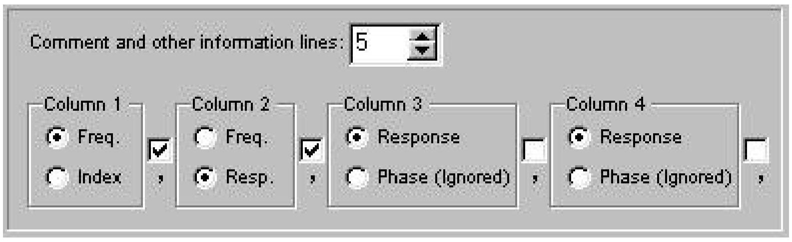
The above figure shows a configuration example for recognizing a microphone compensation file in the following format:

```
MDF (Microphone Data Format) File
Author=LinearX Systems Inc
Date=May 3,2000 Wed 11:52AM
Model=M53
Serial=105271
dBspl= 94.00
dBm= -14.16
Points=552
Index Freq(Hz) dB Deg <-- 9 comment lines
1 10.15 -13.44 0.00
2 10.31 -12.81 0.00
3 10.46 -12.50 0.00
4 10.62 -12.31 0.00
5 10.78 -12.01 0.00
6 10.94 -11.64 0.00
7 11.11 -11.41 0.00
[...]
```

NOTE: to use spectrum data exported from Sample Champion as Microphone Compensation files, select the following settings:

- FFT Size 1K points
- Export frequency data as a TXT file and use the following exporting options (**Settings / File Options**):

- Use the following Manufacturer Microphone Data File Format options:

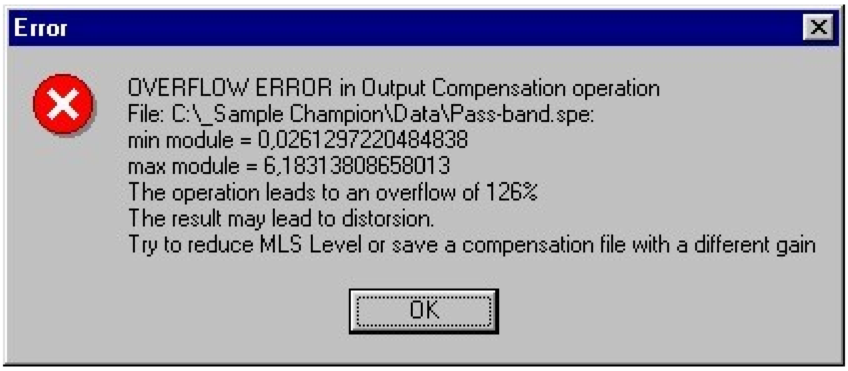


When the **Frequency Weighting** option is enabled, the content of all frequency domain banks (FFT of a selection of scope or impulse response data) are processed by performing the selected frequency weighting (A, C, U or AU).

When the **MLS Equalization (OUTPUT Correction)** is enabled, the output signal is filtered before being sent to the loudspeaker. The selected operation (**Multiply** or **Divide**) is performed in the frequency domain using the selected file (**.SPE**) and complex numbers. This operation is denoted In the flow chart as **OP2**.

The number of samples of the selected frequency **.SPE** file and the internal current **FFT size** must be the same. Also the **Sampling Frequency** of the selected file and the current **internal Sampling Frequency** must be the same.

Since the result of the filtering operation will feed the sound card, it must be smaller than 32767 (the maximum 16 bit integer value) in absolute value, when 16 bit output signal is used. Depending on the filter gain (at every frequency) this could not occur, and cause distortion. In this case the program will show an alert window with some suggestions for solving this problem:

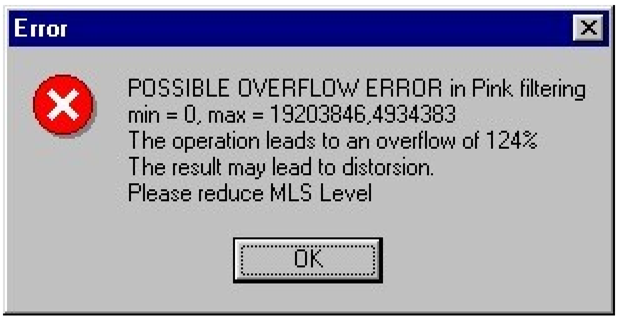


Note 1: filtering is performed by working with complex numbers and the result is converted to long integers only at the end of the process.

When the **Normalize MLS signal** option is enabled, the output signal is normalized. This option **must** be used when the alert window above is shown or when the generated signal is too low or not audible.

A high-precision narrow-band **Pink Filter** can be applied, if desired, to the output signal by marking the appropriate checkbox.

Since the result of the filtering operation will feed the sound card, it must be smaller than 32767 (the maximum 16 bit integer value) in absolute value. Depending mainly on the MLS sequence length and amplitude, this could not occur, and cause distortion. In this case the program will show an alert window with some suggestions for solving this problem:



Note 1: filtering is performed working with complex numbers and the result is converted to long integers only at the end of the process.

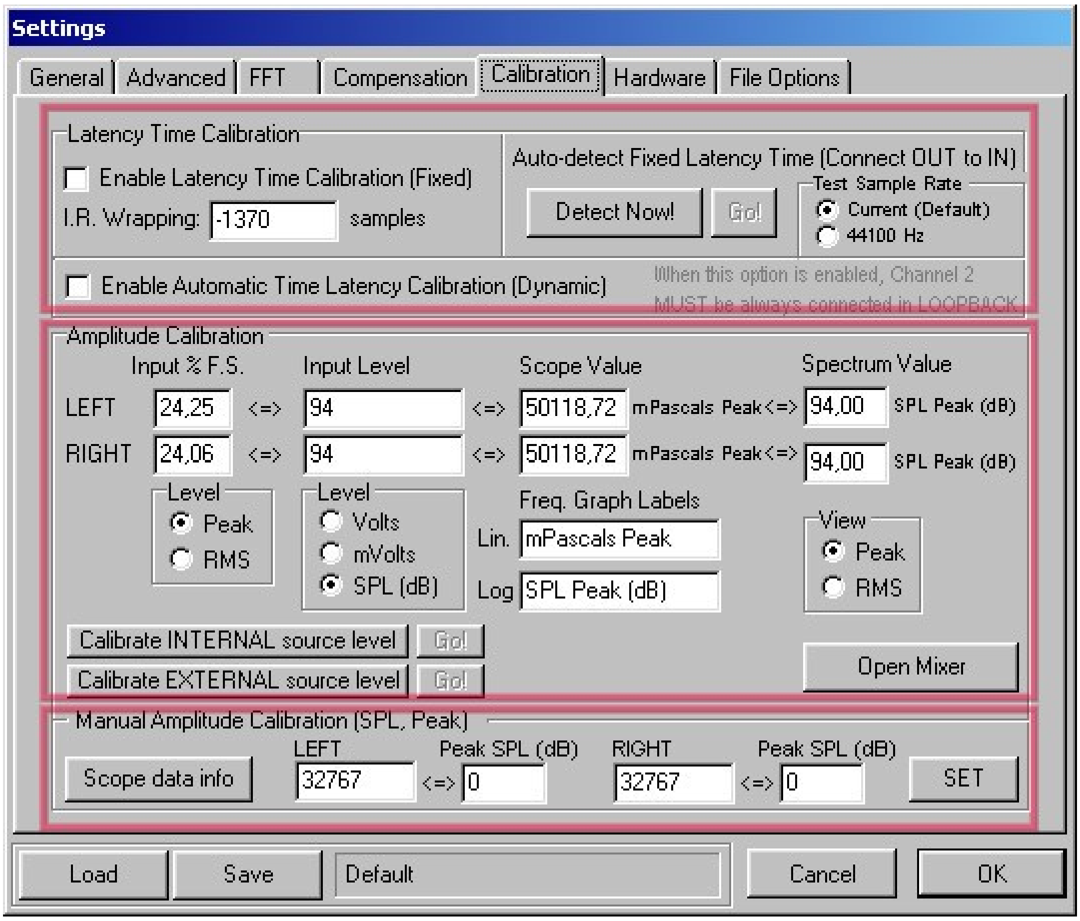
Note 2: the min and max values written in the alert window depend also on the current calibration.

The **Free Field Reference Impulse Response** selection allows the selection of a **.IRE** file that can be plotted (overlapped) in the Impulse Response Window. This feature could also be used by future developments of the program or by specific plugins. The Impulse Response in the file must use the same Sampling Frequency inserted in the current program settings.

The **Default Dir** button allows the selection of a default directory for file operations.

Settings - Calibration

 Settings - Calibration



Different calibration types are available: Latency Time (Fixed and Automatic) and Amplitude Calibration (Real-time and Manual).

Latency Time Calibration:

it is the delay (due to soundcard hardware and drivers and operating system) between input and output buffers; it depends on the specific audio card that is used. The effect of this delay is a left or right shifting of the Impulse Response peak (ideally placed at t = 0 for a loopback mesurement).

Sample Champion allows 2 ways to compensate the soundcard latency: **FIXED** and **AUTOMATIC / DYNAMIC** (recommended).

NOTE: the **FIXED** calibration can be used when the soundcard presents always the same latency delay every time that a measurement is started (**SYNC REC/PLAY**).
A soundcard has fixed latency if the peak of a measured loopback impulse response is placed always at the same time position, in different measurements.

If, in different measurements, the peak of a measured loopback impulse response is placed at different time positions, the **DYNAMIC** latency calibration option must be used. In this last case the measurement must be stereo and the channel 2 must be in loopback (soundcard output channel 2 connected to soundcard input channel 2).

Generally the **DYNAMIC** latency calibration option is preferred when external soundcards (USB or Firewire) are used.

LATENCY TIME CALIBRATION PROCEDURE (FIXED)

Automatic procedure: press the **Detect Now!** button and the **Go!** Button. The delay is measured and set automatically. Note that the **INPUT** of the audio card must be directly connected to its **OUTPUT** and the levels must be correctly set (see the manual procedure below for more details).

Some audio cards are not compatible with the automatic procedure. In this case, the manual procedure can be used.

Manual procedure:

- Connect the INPUT of the audio card directly to its OUTPUT (loop back)
- Open the Impulse Response Window
- Open the Settings Window and set (for example) the following parameters:
- In and Out 16 bit, Mono
- Buffer Length = 4K
- Block = 1
- Mode: Repeat
- Sampling Rate 44100 Hz
- Step: 2
- MLS: 4K (12, 11, 10, 2)
- Uncheck any Output Correction
- Press **Syncro Start/Stop** and perform a measurement

- If the input level is too low, open the mixer and adjust it

- Locate the X value of the impulse response peak by moving the cursor on the wave (use the **Locate Peak** button and the **Show Info on Image** for help)

- Open the Settings|Calibration window and insert the X value found in the **Latency Time Calibration** field, changing its sign, and check **Manual** (for example if X=112, write: -112)

- Repeat the single measure procedure and search for the peak. Now it should be located near zero (the *t=0* value is not visible). If this does not happen, adjust the latency time value until the peak is in the proximity of zero. Remember that the Impulse Response recovered using MLS is wrapped at the end of the buffer length: an excessively negative value will cause the impulse peak to move to the end of the data buffer. In this case simply increase the inserted value as much as necessary.

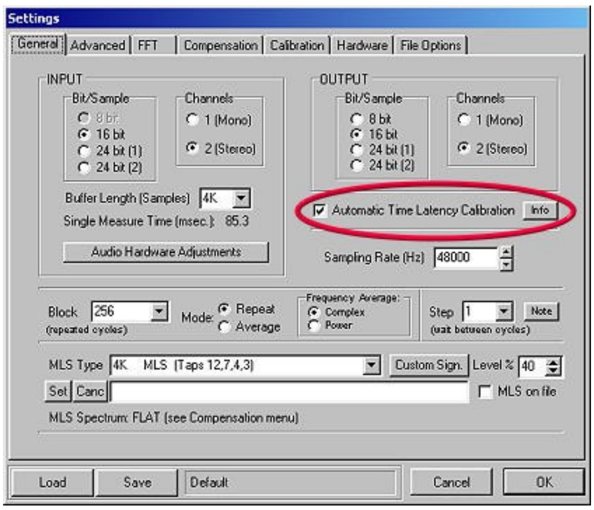
Note: every audiocard has a specific latency time value. Just one measurement is needed. The latency time value could change when audio card drivers or configuration are updated and (in rare cases) changing sample frequency.

AUTOMATIC LATENCY TIME CALIBRATION (DYNAMIC)

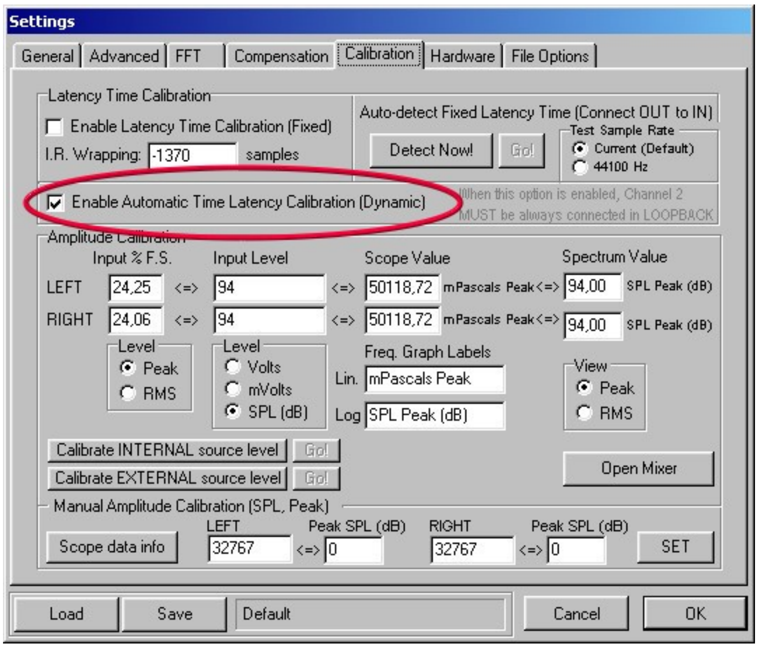
This option allows a very robust synchronization of your soundcard and avoids all errors due to data sync loss. It allows extremely accurate measurements when used in the Average mode.

When the auto-sync function is active, the time latency calibration (that depends on many factors: soundcard hardware and drivers, operating system, etc..) will be automatically compensated.

It can be enabled in the Settings/General window:

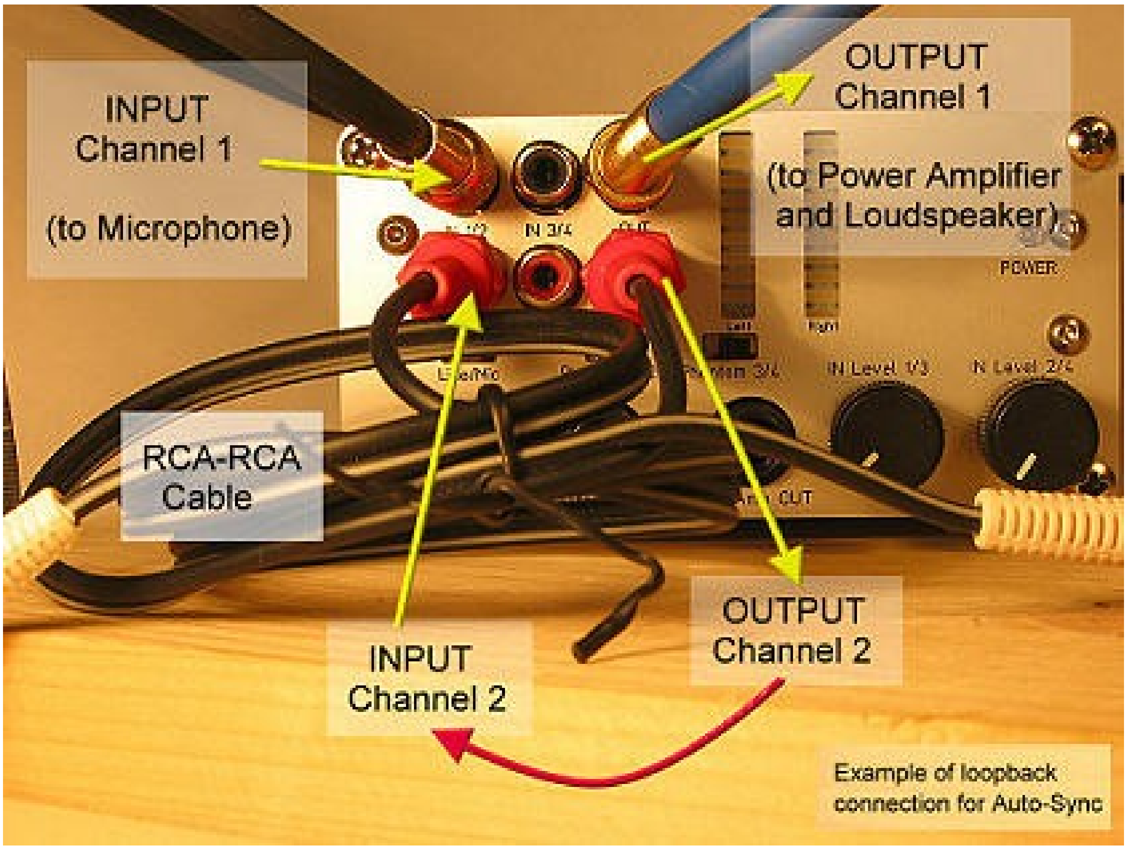


or in the Settings/Calibration window:

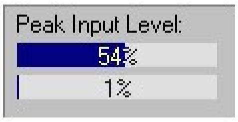


When this option is enabled, the soundcard **Channel 2 INPUT** must be connected directly to **Channel 2 OUTPUT** (loopback).

Example of connection:



Adjust input and output levels of Channel 2 in order to obtain a low input peak level. Usually a value of 1% or 2% of the peak level is enough to activate the sync function.



The **microphone** (or any other input device) can be connected to **Channel 1 INPUT**. The power amplifier and **loudspeaker** can be connected to **Channel 1 OUTPUT**.

When this option is active, the saving window consents to optionally save only the impulse response channel 1 data (channel 2 contains only sync information).

Tip: In most of soundcards the crosstalk between channels and between in / out is sufficient for the sync information (very sensible). In this case the Automatic latency feature can work also without connecting a loopback cable on channel 2.

Amplitude Calibration:

it is possible to perform an amplitude calibration in real-time, using the internal signal generator or an external calibrator (standard 1 kHz calibrator), or manually.

- The **Input % F.S.** field shows the measured input peaks (Left and Right channel) as full scale percentage

- The **Input Level** field must be used to insert the requested value (be sure to use '.' or ',' according with the international settings of your operating system)

- The **Scope Value** and **Spectrum Value** fields are filled by the program at the end of the calibration procedure

- All other check buttons can be selected by the user to change the input calibration level mode (Peak or RMS), the level units (Volts, mVolts or SPL (dB)) and the view mode (Peak or RMS)

AMPLITUDE CALIBRATION PROCEDURE

CALIBRATION USING AN INTERNAL SOURCE:

- Open the Settings|Calibration window

- Connect the INPUT of the audio card directly to its OUTPUT (loop back)

- Press the **calibration using an INTERNAL source** button to start the procedure. A pure tone (1 kHz) will be generated by the sound card

- Open the mixer by pressing the **Open Mixer** button, check by means of the input peak level meter (if available) the absence of any saturation and that the indicated values are reasonable (for example -6 dB for both channels). Check also that the output peak level meter (if available) does not indicate any saturation and that the indicated values are reasonable

- Write in the **Input Level** fields the desired values for left (or mono) and right channels. Select also all other options (Peak, RMS...) as desired

- Press the **Go!** button and wait the ENDED status of the sampling engine

- The **Scope Value** and **Spectrum Value** fields should be now filled by the program with the requested values. From now on, all measured values will be calibrated according to these values

CALIBRATION USING AN EXTERNAL SOURCE:

- Open the Settings|Calibration window

- Place on the microphone an external calibrator and turn it on

- Press the **calibration using an EXTERNAL source** button to start the procedure

- Open the mixer by pressing the **Open Mixer** button, check by means of the input peak level meter (if available) the absence of any saturation and that the indicated values are reasonable (for example -6 dB for both channels).

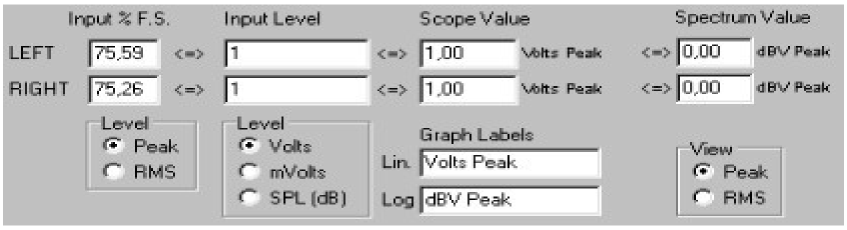
- Write in the **Input Level** fields the desired values for left (or mono) and right channels. Select also all other options (Peak, RMS...) as desired

- Press the **Go!** button and wait the ENDED status of the sampling engine

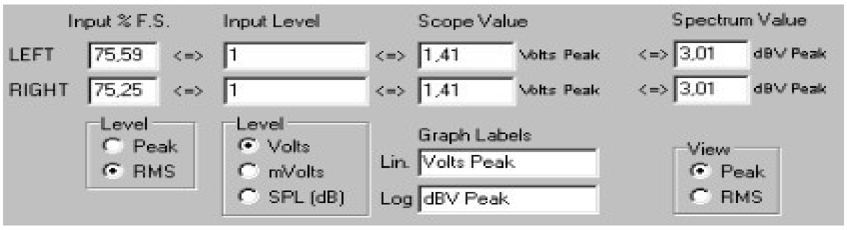
- The **Scope Value** and **Spectrum Value** fields should be now filled by the program with the requested values. From now on, all measured values will be calibrated according to these values

Examples of calibraton:

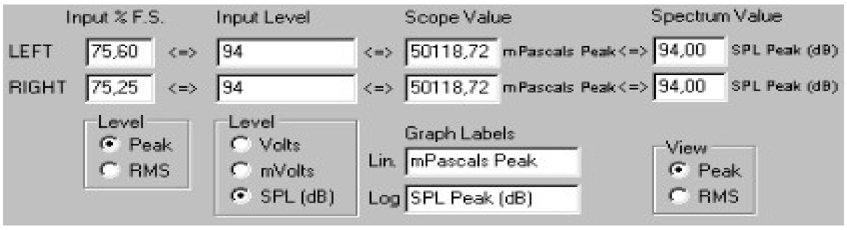
(1)



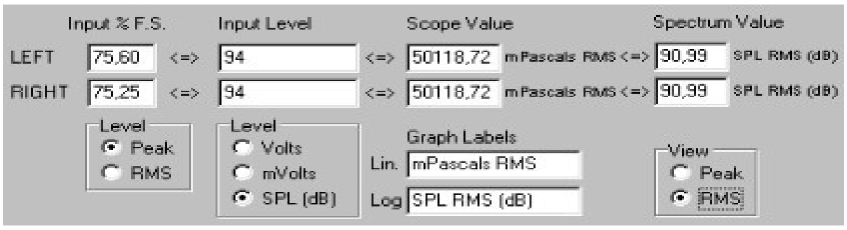
(2)



(3)

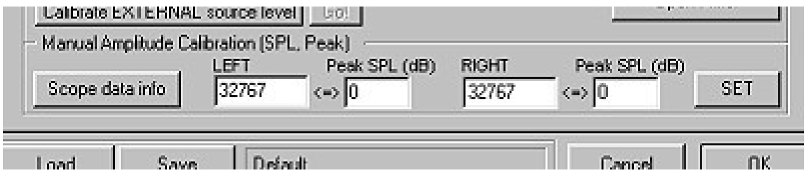


(5)



MANUAL CALIBRATION:

In the Setting / Calibration window the following section is present:

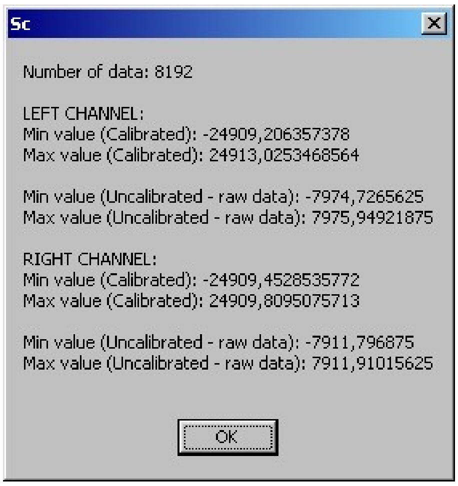


The **manual procedure** allows to set a dB (Peak) calibration without a real measurement. The user can assign to a time peak value (Scope window, raw sampled data) a dB (Peak) power level by pressing the **SET** button.

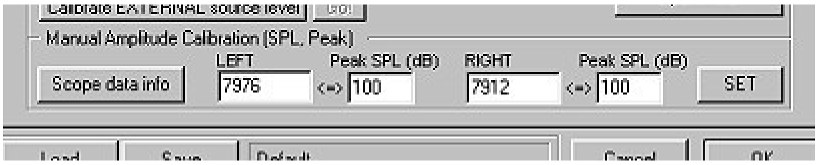
In the example above a peak value of 32767 in Scope window has been assigned to **0 dB**.

In this way a 16 bit WAV file with a sine signal between +/- 32767 will show a spectrum with a peak of 0 dB. With this same calibration a WAV file with a sine between +/-327 will show a peak in the spectrum of about -40 dB.

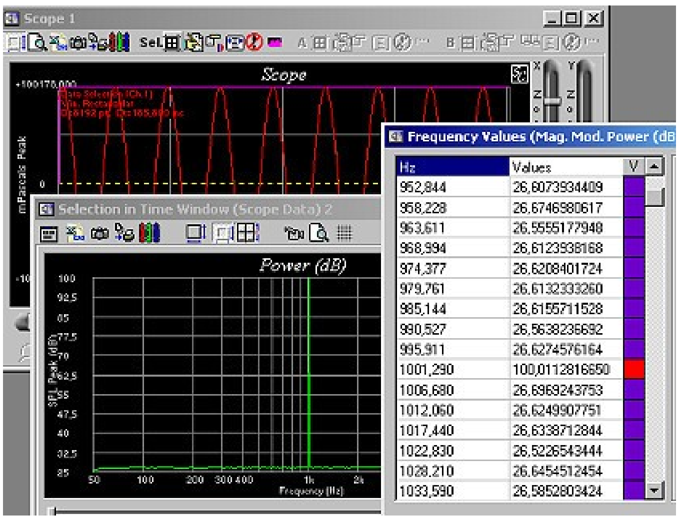
The **Scope data info** button searches minimum and maximum values of the current data in Scope buffer (loaded from a file or measured in real-time). The **uncalibrated** maximum data (Left and Right channels) are automatically written in the LEFT and RIGHT fields of manual calibration.



The manual calibration is very simple: after pressing the **Scope data info** button, the user can write the desired dB value for LEFT and RIGHT channels (only LEFT for mono) and press **SET**. This procedure assumes that a pure tone signal is in Scope data buffer.

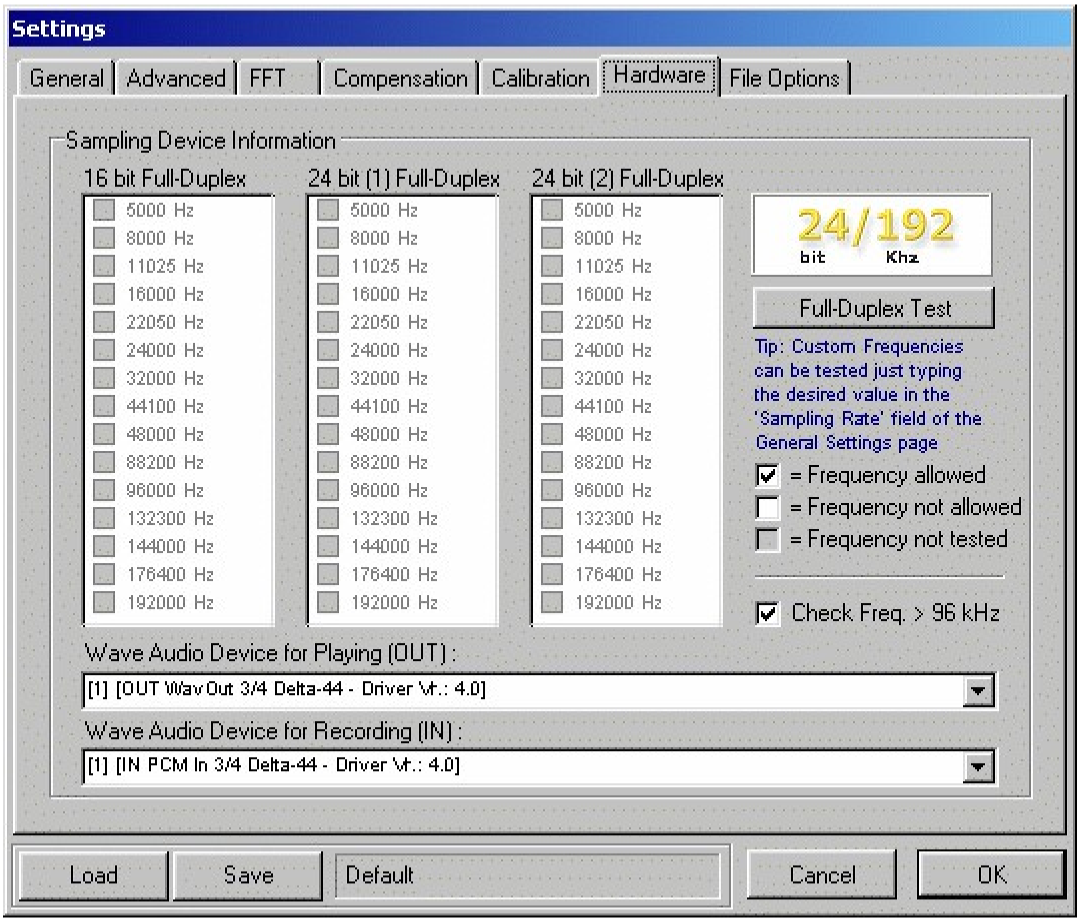


In the above example a 100 dB value has been set. The LEFT and RIGHT values (7976 and 7912) are the **raw** (uncalibrated) max values found in the Scope window, written automatically after pressing the Scope data info button.



Settings - Hardware

Settings – Hardware

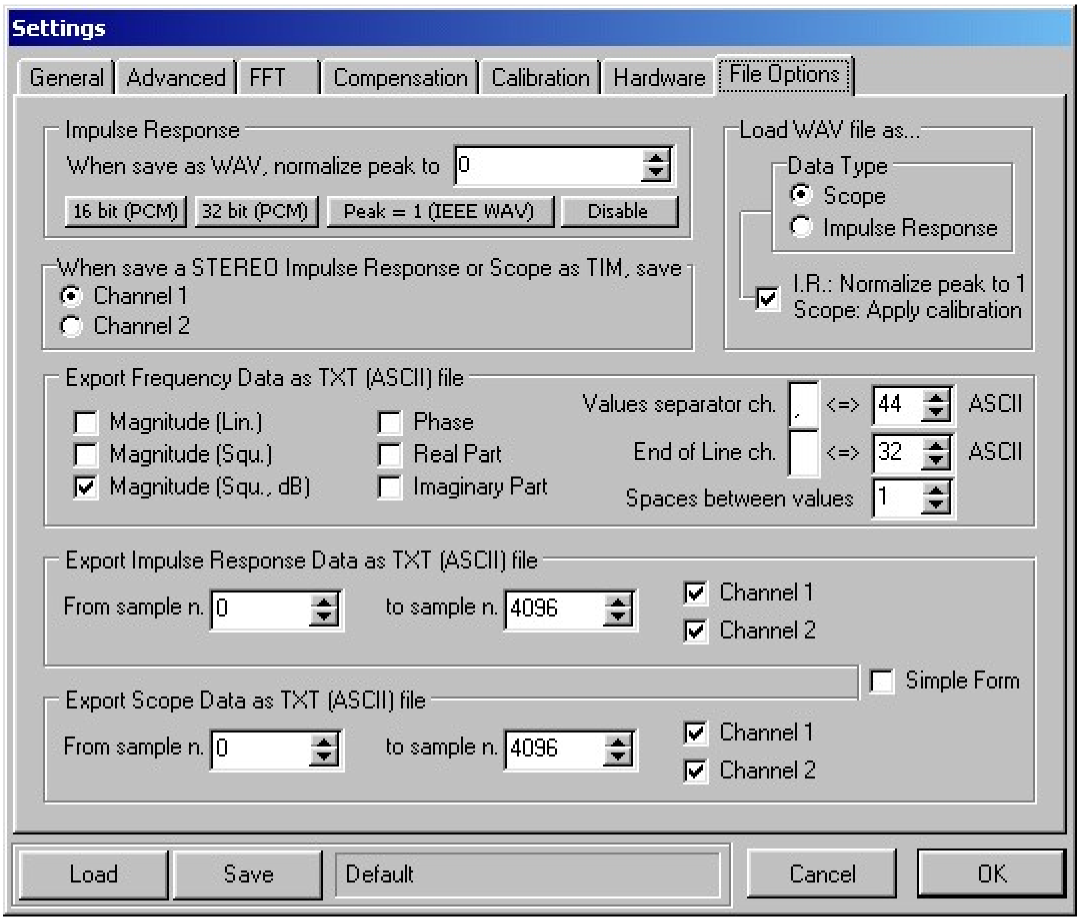


Here some information about the installed sound card(s) are shown.

Note that in Windows Millenium, Windows 2000 and Windows XP some frequencies not actually supported could be shown as available (the operating system emulates those frequencies).

Settings – File Options

 Settings – File Options

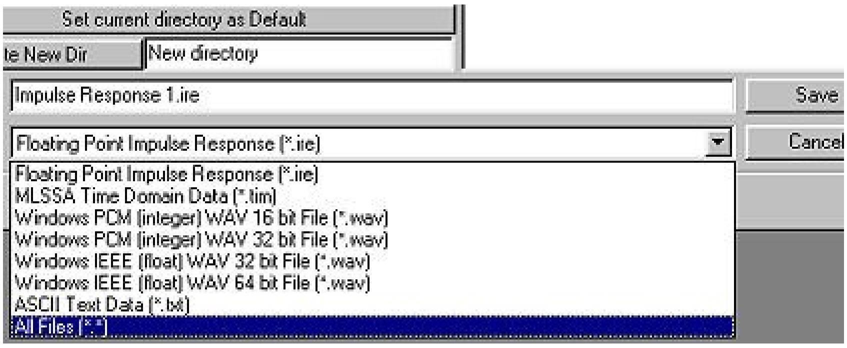


Save Impulse Response as WAV

Impulse Response can be exported as WAV file. The following WAV formats are supported:

PCM WAV 16 bit (Integer format)
PCM WAV 32 bit (Integer format)
IEEE WAV 32 bit (Floating Point format)
IEEE WAV 64 bit (Floating Point format)

The WAV type can be selected in the drop-down menu of the Save window.



Some normalization options concerning WAV files LOAD and SAVE are available.

Example: a PCM 16 bit WAV file can handle integer data between +/-32767. A measured impulse response has a peak value of 0.254 mPascal. If it is saved as PCM 16 bit WAV without normalization, the resulting WAV file will contain only zero values.

The same happens when saving as PCM 32 bit WAV, that can handle integer data between +/-2147483647.

The IEEE floating point WAV format does not have this problem since can handle floating point data. A normalization to "1" is available anyway for giving the better compatibility with other software.

The normalization feature can be disabled setting the normalization peak value to "0".

TIM File Exporting Options

When a measured **STEREO** IR or Scope is saved in **.TIM** format, it is possible here to select the channel to be exported (TIM files support only 1 channel).

Load WAV File as...

A WAV file, when loaded, can be recognized by Sample Champion as **Scope** or **Impulse Response** Data.
When loading a WAV file (that can be classified as Scope or Impulse Response by the Load WAV file as setting), another option is available:

- Impulse Responses: the peak can be optionally normalized to "1"

- Scope data: the current amplitude calibration can be optionally applied.

Frequency Data Exporting Options

Some options for exporting Fequency Data as ASCII file (type of exported data, separator characters, spaces between values) can be selected here.

Impulse Response and Scope Data Exporting Options

Some options for exporting Impulse and Scope Response Data can be selected here.
The **Simple Form** exporting option enables the simplest format of the exported Impulse Response and Scope Data.

ASCII (TXT) Impulse Response files measured by using other software can be also imported. Only single channel Data are accepted. The format of the imported files is the following:

Row nr. 1 --> Title (Text, Ignored)
Row nr. 2 --> DeltaT (ms)
Row nr. 3 --> Number of Data = N
Row nr. 4 --> Data nr.1
Row nr. 5 --> Data nr.2
Row nr. 6 --> ..

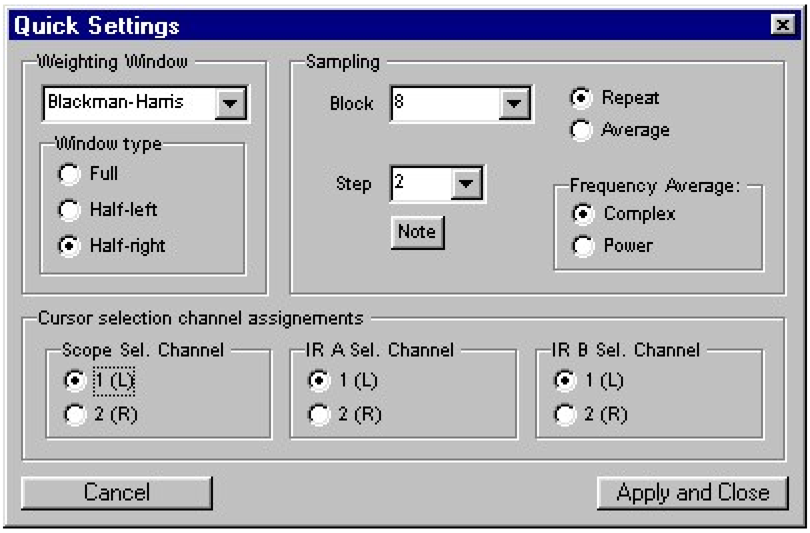
Last Row ---> Data nr.N

Example (4096 data, Fs=48kHz):

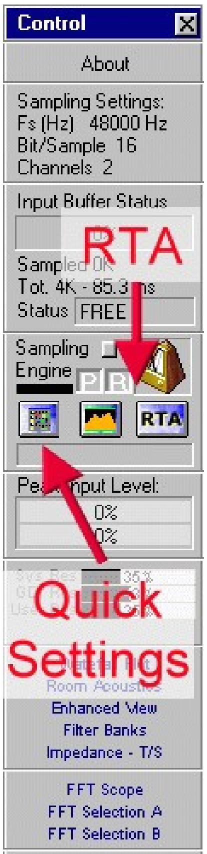
My Impulse Response
0.0208333 **<-- This is DeltaT (ms); DeltaT=1000/Sampling Frequency**
4096 **<-- This is the number of the following Data**
0.00114076 **<-- This is the Data nr. 1**
0.00135692 **<-- This is the Data nr. 2**
...

The above format is similar to the format of the files exported by the MLSSA™system. IR and Scope Data are exported by Sample Champion in this format when the *Simple Form* option is enabled.

Quick Settings

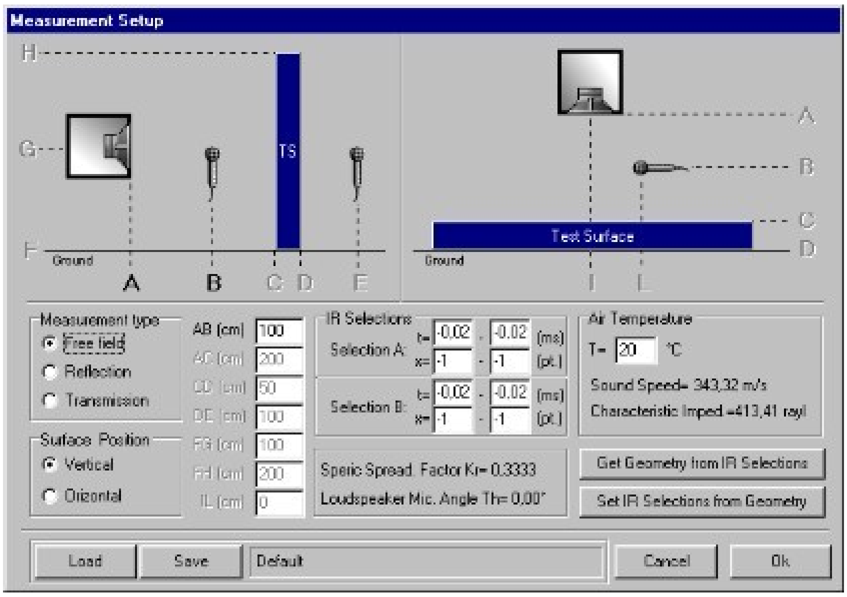


This window can be opened by clicking the Menu Options/Quick Settings item or the button on the control bar. When plugins are activated, the Settings can no longer be changed. Using this function, however, some of the settings can be changed also in this case.



Measurement Setup

 **Measurement Setup**

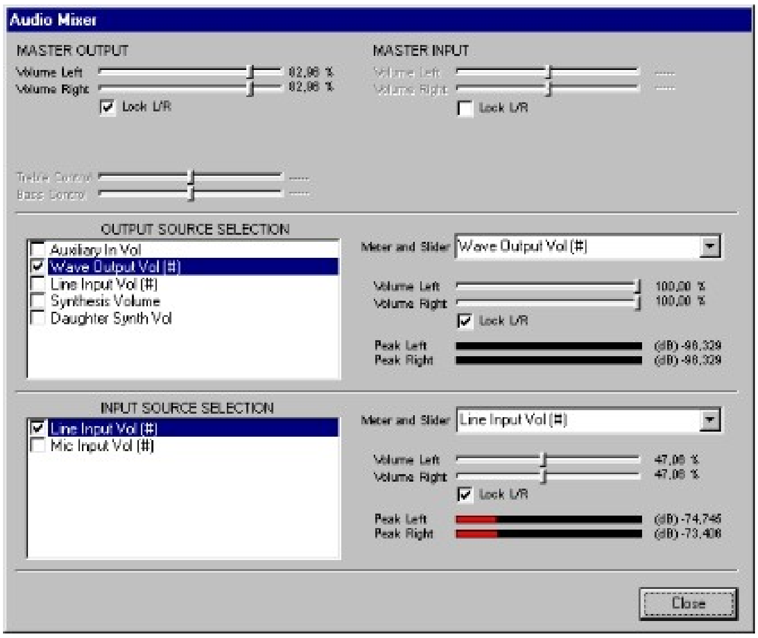


- In this window the user can set or edit some geometry parameters, and **Save** them in an **.INI** file.

- It is possible also to **Load** a set of parameters previously saved in a **.INI** file. The current version of the program uses only the information concerning **Selection A** and **Selection B.**

Mixer

Mixer



The built-in mixer is a general purpose mixer. It tries to read all input and output sources and the available real-time peak meters from the sound card driver. Some hardware is not completely compatible with the internal mixer. In these cases please use the standard Windows™ mixer or the mixer application provided by the audio card manufacturer. See **Tweaks** function to find more information about selecting the mixer application.

The **OUTPUT SOURCE SELECTION** list shows all available sound card outputs. If a source is **checked** the corresponding output is **enabled**.

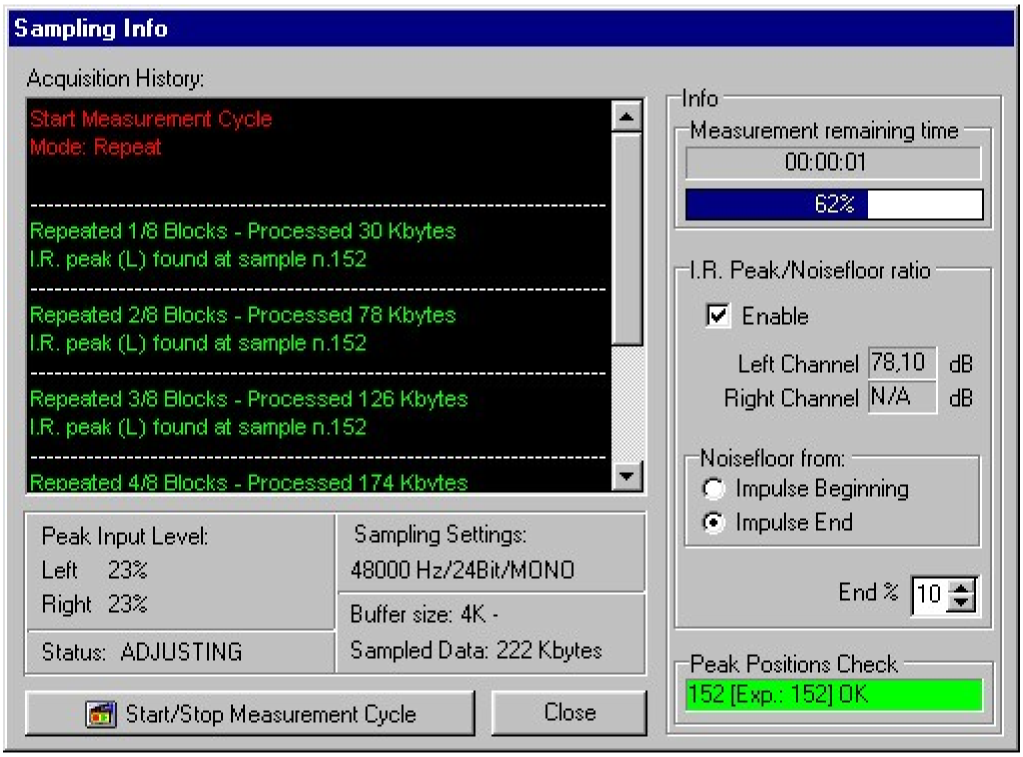
The **INPUT SOURCE SELECTION** list shows all available sound card inputs. If a source is **checked** the corresponding input is **enabled**.

It is recommended to select **ONLY** the input (line or mic) and the output (wave out) sources strictly necessary for the measure (see for example the picture above, concerning a Turtle Beach Pinnacle audio card).

Generally speaking, it is better to use the Line In input instead of the Microphone input because Line In has a better signal/noise ratio. There are, however, some positive exceptions (for example the TB Pinnacle audio card, that can handle also condenser microphones with phantom power).

Measurement Info Window

Measurement Information



This window shows some information about the measure. At every measurement cycle, the number of blocks that have been processed and the peak position are reported here. This log should be used especially for Impulse Response measurements: the value of the peak position **must** be the same for every measurement in the same cycle. If this does not happen, some measure error has occurred and further investigation is required.

Other features in the Information window:

- Estimated remaining measurement time

- Impulse Response Peak/Noise ratio

- Automatic control of repeated measurement correctness, checking peak position of every measure

The Information window can be (optionally) opened when a measurement starts (from the program menu and the floating menu it is possible to enable the automatic management of this feature).

During measurements in **Blinded Mode** this is the only visible window and the remaining parts of the screen are black.

During normal measurements this window can be open or closed.

It is possible to select all or part of the history log using the mouse and copy it to the clipboard using Control-C.

Show/Hide Control Bar

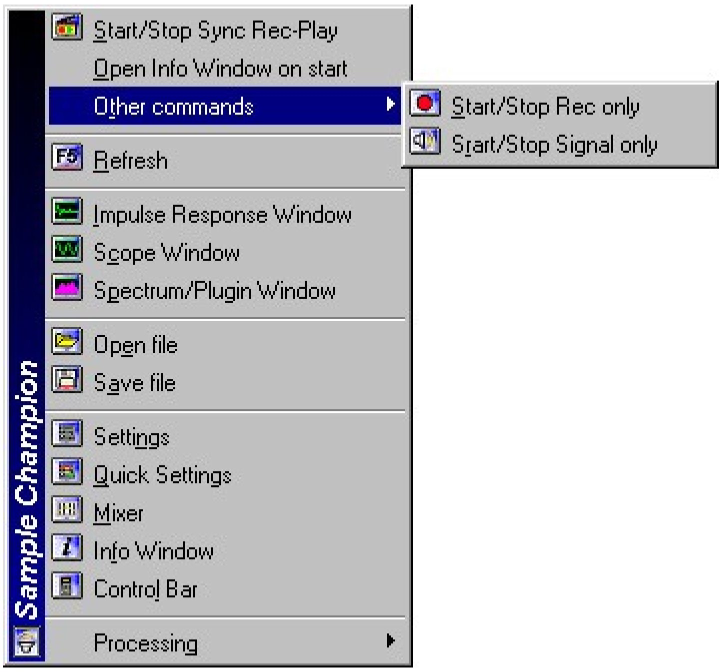
Show/Hide Control Bar

This button can show or hide the control bar on the right of the screen. To reposition the control bar at its original position, move it outside of the screen on the right and press twice this button.

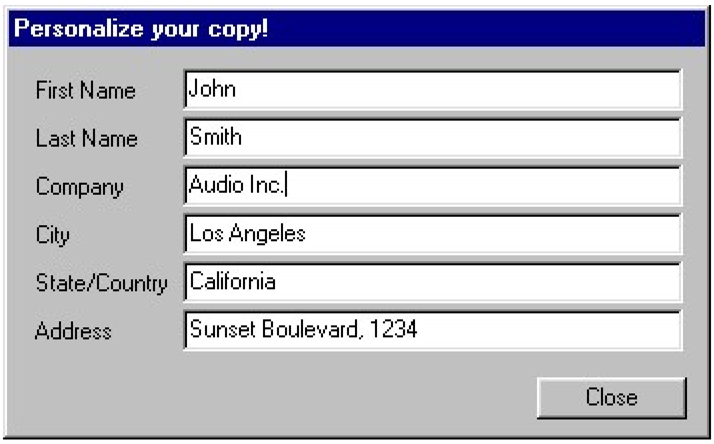
The control bar are shows some important information on the current measurement parameters and state.

Floating Menu

The most important menu commands are available in this menu, Right-click on the **program desktop** or on the **control bar** to open this menu.

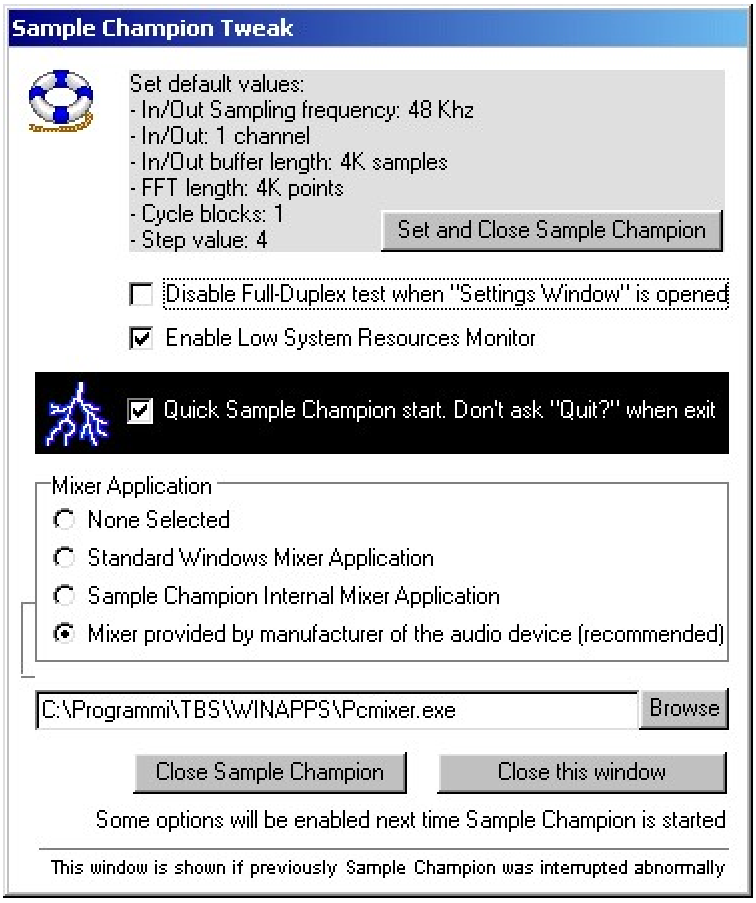


Personalization



The program can be personalized with Name, Address...

Tweaks

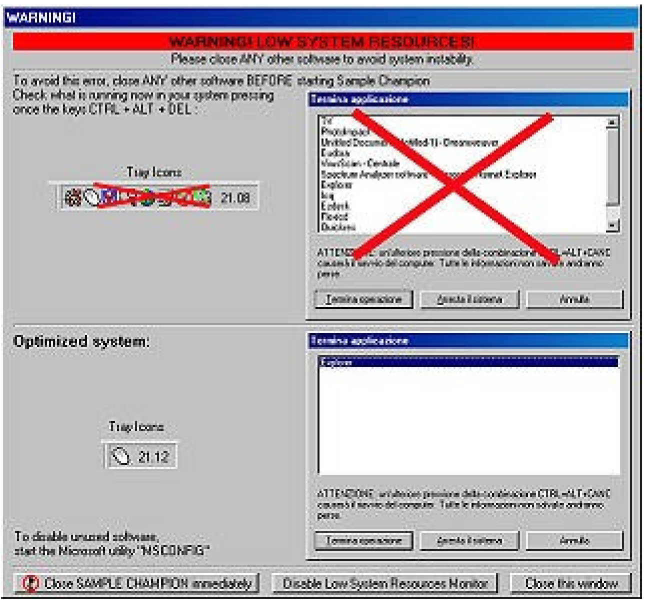


- Here some ***emergency*** and ***service*** functions can be accessed. The selection of the mixer application is located here.

- In some rare cases the Full-duplex test performed when the Settings window is opened causes troubles and errors. This test can be disabled from this menu.

- Under Windows 98 it is available a global system resource monitor, integrated in Sample Champion. The resource usage level information is located on the remote bar.

When system resources are too low, a warning window is opened, and some actions are suggested :

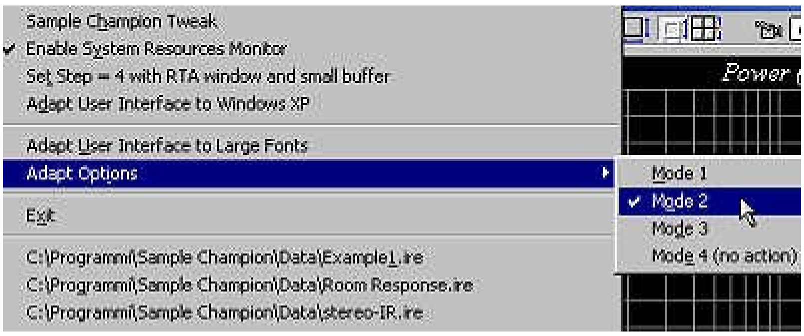


This function can be enabled only in Windows 95, 98 and Me.

Other *fine tuning* options are available (File menu):

- **User Graphical Interface adapting to XP**. This can be enabled when SC runs in XP family operating systems to adapt the windows sizes and other graphic parameters.

- **Adapt to Large Fonts**: Sample Champion can work correctly also when Large Fonts are in use in Windows. Since the Windows appearance, with Large Fonts in use, is different in different Windows versions, the user can select the better Mode, for an optimal setting.



The toolbar will be shown correctly when the optimal Mode is selected. The Adapt to Large Fonts function is enabled or disabled automatically when the user changes the Font setting in windows, but can be also enabled or disabled manually. The Large/Small Fonts setting in Windows can be found in Screen/Properties/Settings/Advanced: Large Fonts (120 dpi) or Small Fonts (96 dpi). The Small Fonts setting is anyway recommended.

Sampling Engine Commands

Record
This button starts the sampling engine **without** starting the signal generator.

Play
This button starts the signal generator **without** starting the sampling engine.

Sync Rec-Play
This button starts the signal generator **and** the sampling engine synchronously.

Windows Managing

Cascade
This button arranges all child windows inside the program in the **Cascade** mode.

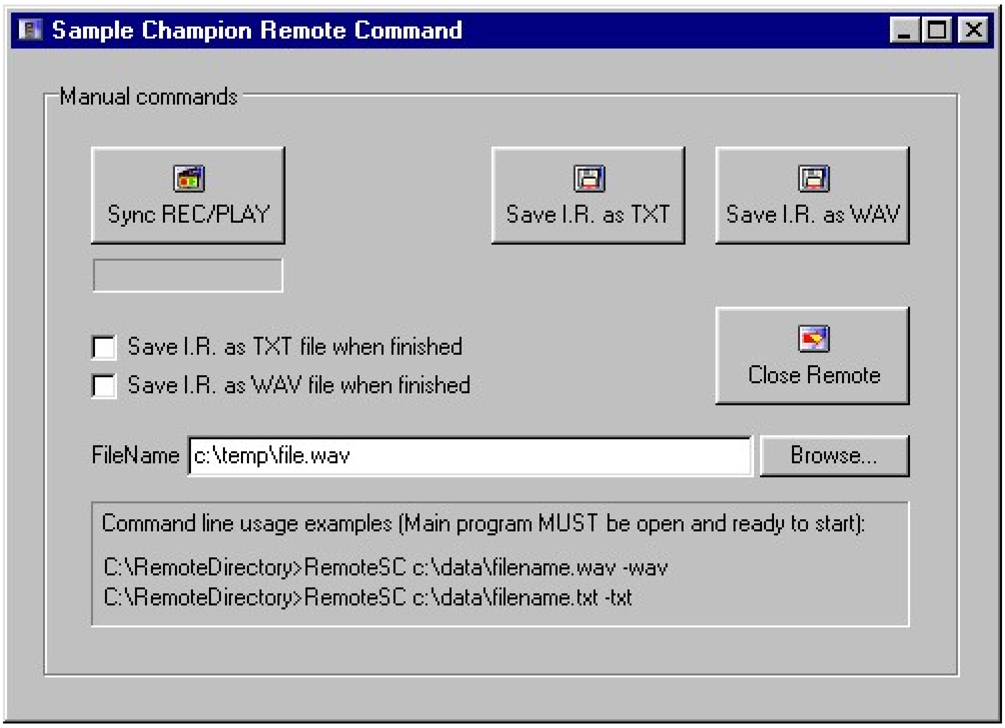
Tile
This button arranges all windows inside the program in the **Tile** mode. It is particularly useful when only 2 windows are open (for example Impulse Response and FFT of Selection A of Impulse Response).

Minimize
This button **Minimize**s all child windows inside the program.

Refresh
This button **Refresh**es the plots in all child windows inside the program.

NOTE: The refresh action has also some side effects such as reprocessing the data in all enabled selections (see previous sections).

RemoteSC



- Sample Champion can now be triggered by means of a **command line** (from a DOS window or an external program). A small utility called **RemoteSC** is included in the setup package. This utility can be called from a command line or from any external program. It allows to start a measurement and save automatically the measured Impulse Response as TXT or WAV files.

BEFORE using this utility:

- Start and leave open Sample Champion that must be ready to start a measurement

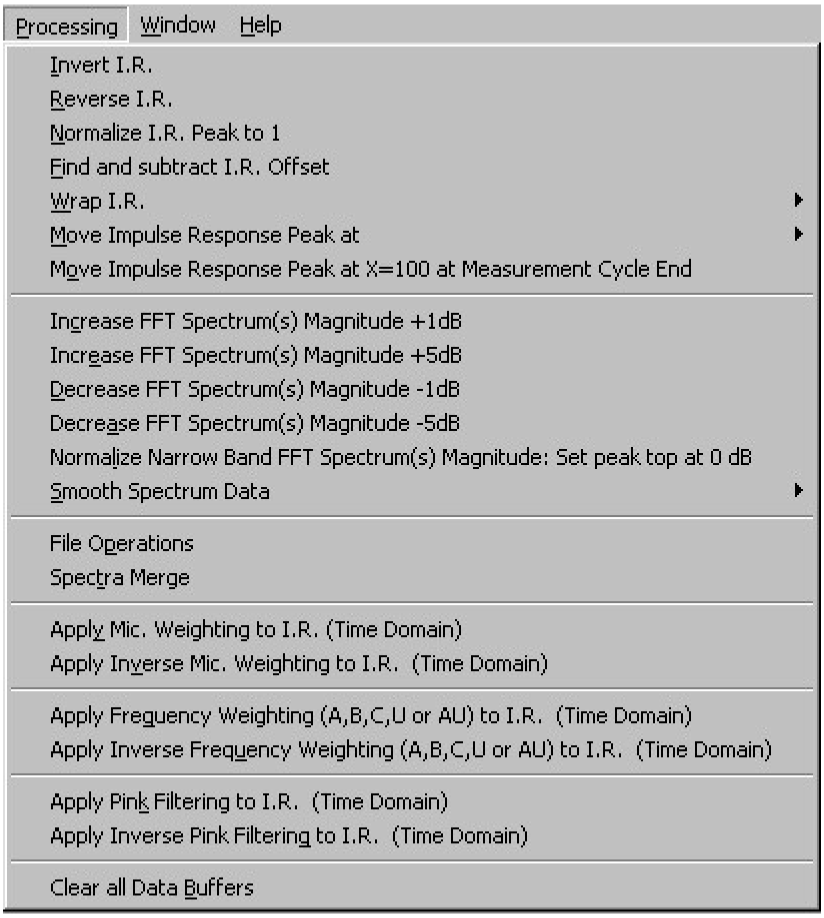
- **Usage example:**

C:\RemoteDirectory>RemoteSC c:\data\filename.wav -wav
Starts a measurement cycle and saves the measured Impulse Response as a WAV file

C:\RemoteDirectory>RemoteSC c:\data\filename.txt -txt
Starts a measurement cycle and saves the measured Impulse Response as a TXT file

From the website it is possible to download an example, which includes the Delphi 5 source code, for calling RemoteSC from a custom program.

Post-Processing Functions



Some useful post processing functions can be accessed from this menu.

Invert I.R.

This function changes the sign of the current I.R.

Reverse I.R.

This function reverses the current I.R. with respect to the middle time point.

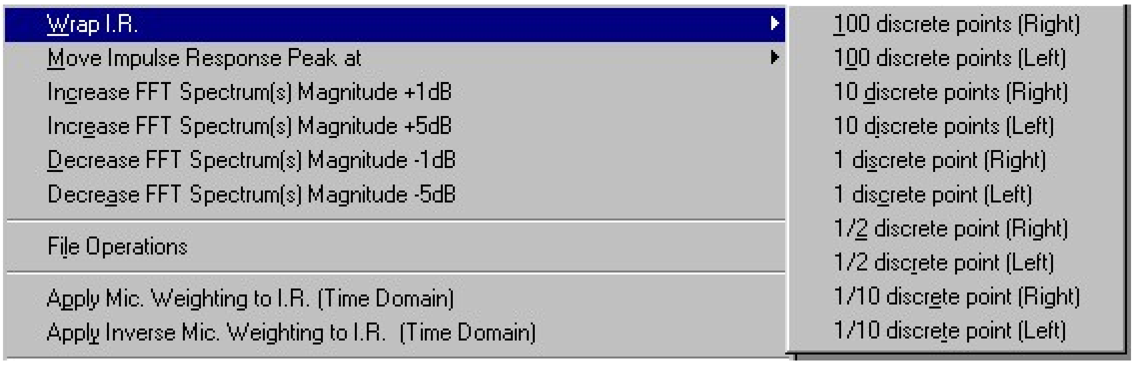
Normalize I.R. Peak to 1

This function normalizes to 1 the peak value of the current I.R..

Find and subtract I.R. Offset

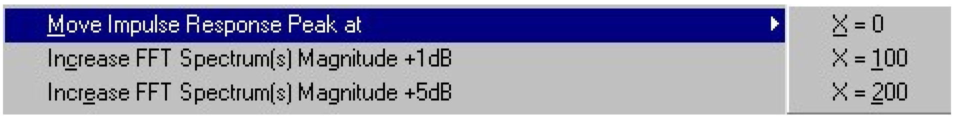
This function finds the continue component and subtracts it from the current I.R. This can be useful also when the "Remove offset from sampled data" option (in Settings Advanced) is enabled, because here the operation is performed on the final I.R. (and not on the raw sampled signal before the computation of the Impulse Response).

Wrap I.R.



These functions wrap the current I.R. by the selected amount of points (also by fractions of discrete points). This can be useful when the Latency Calibration Time had not been set correctly at measurement time. Note that an I.R. measured with the MLS method has the same period as the MLS, so wrapping doesn't generate discontinuities.

Move Impulse Response Peak at...



These functions are similar to previous **Wrap** functions. The peak can be moved at the selected (fixed) X position.

Move Impulse Response Peak at X=100 at Measurement Cycle End

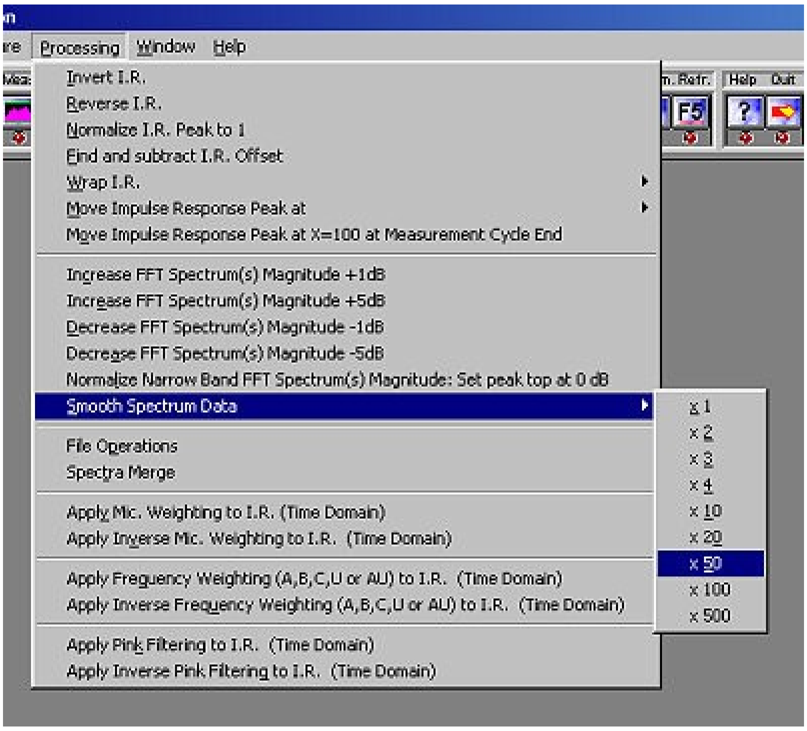
When enabled, this option automatically place the impulse peak at sample N. 100 when the measurement cycle is ended.

Increase / Decrease / Normalize FFT Spectrum(s) Magnitude

These functions can be useful for a quick comparision of frequency spectra measured at different amplitude levels. See also **Edit Frequency** function

Smooth Spectrum Data

This is a post-processing function for smoothing spectra. The number of cycles can be selected, to obtain a more deep effect.



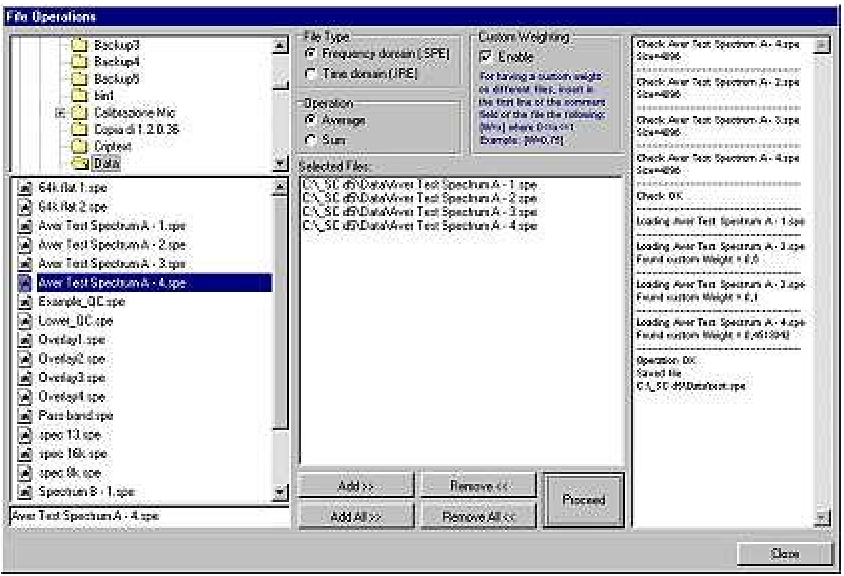
The smoothing is obtained performing a moving average on the frequency data. The spectrum, after the smoothing, will have only Real data (the Imaginary part is set to 0). The original (unsmoothed) spectrum can be restored by pressing F5 (refresh).

File Operations

Frequency files (**.SPE**) and Impulse Response files (**.IRE**) can be summed or averaged.
If the first line of the comment field of the file has the following content:

[W=x] where $0 \leq x \leq 1$,

the file is weighted according to the specified value.



For example **[W=0,5]** attributes to the file a 0,5 weight. This can be useful for performing loudspeaker sound power measurements or spatial averages of measurements made in different points of a concert hall.

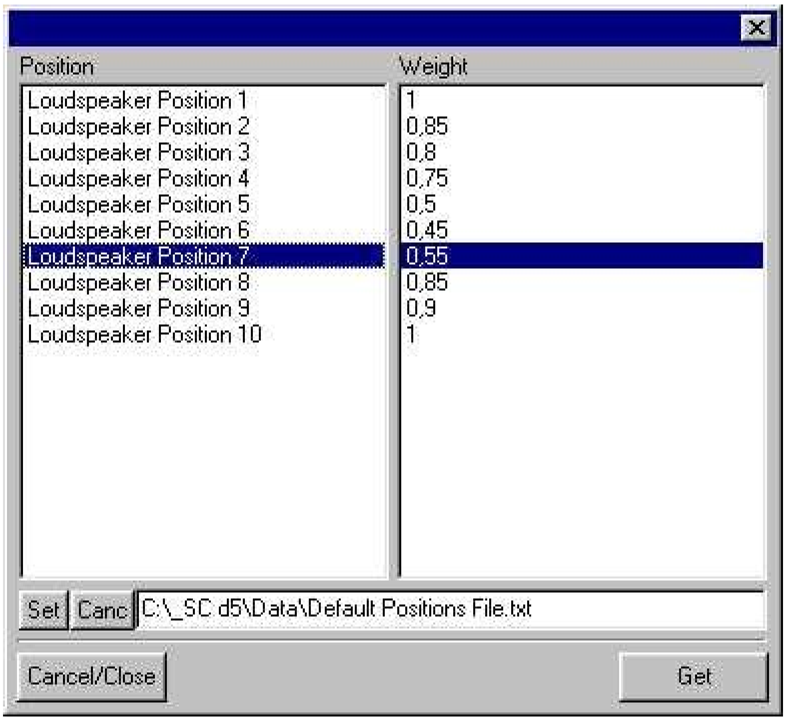
The File Operations function can be used also for **rescaling** a frequency or time file by any desired factor (just select a single file, with the desired rescaling factor in the comment field as described above).

A custom **TXT file** with a list of **Position-Weighting associations** can be created by the user. The desired weighting value can then be obtained by pressing the button **Get Weight from File** in the Save Dialog before saving the file. The TXT file must contain a list of weights in this form:

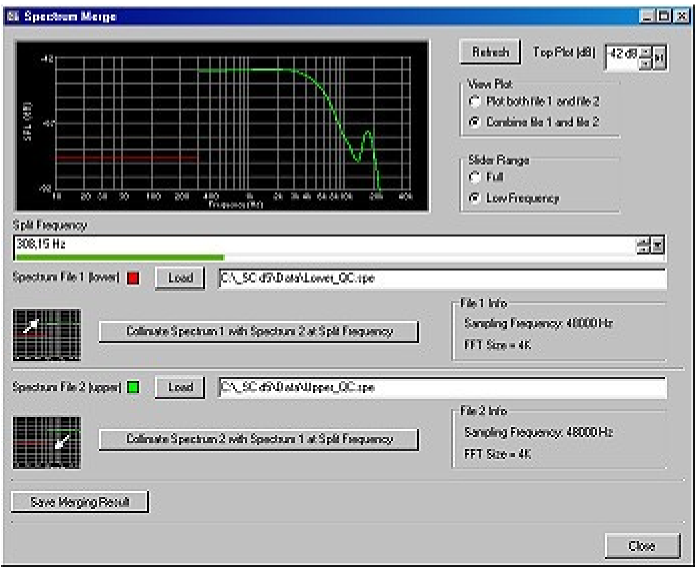
[Loudspeaker Position 1]=[1]
[Loudspeaker Position 2]=[0,85]

[Loudspeaker Position 3]=[0,80]
[Loudspeaker Position 4]=[0,75]
[Loudspeaker Position 5]=[0,5]
[Loudspeaker Position 6]=[0,45]
[Loudspeaker Position 7]=[0,55]
[Loudspeaker Position 8]=[0,85]
[Loudspeaker Position 9]=[0,90]
[Loudspeaker Position 10]=[1]

A .TXT file with these values will generate a **Position-Weighting associations** list as shown in the following picture:



Spectra Merge



This function is very useful for merging **Nearfield** with **Farfield** measurements. This function can be activated from the Processing menu.

This is an **example** concerning the use of this function (with 2 of the spectrum files in the setup package):

- Open SC and select in the processing menu "Spectra merge"

- Load as File 1 "Lower_QC.spe"

- Load as File 2 "Upper_QC.spe"

- Select "Combine file 1 and 2" and press Refresh (set Top Plot value around -42 dB for plotting the spectrum)

- Now you can "attach" the first spectrum to the second at the split point (or vice versa) and save the result of the operation

Apply Mic. Weighting (and Inverse) to I.R.

This function applies (or removes, if inverse) the current microphone compensation to the current I.R. Note that when Microphone Compensation (Settings/Compensation) is enabled, only the FFT data are compensated). This time data filtering operation requires remarkable computational resources and cannot be performed in real-time. It can be useful when an Impulse Response must be processed by a Time Domain plug-in (Waterfall Plot, Room Acoustics, Enhanced View...).

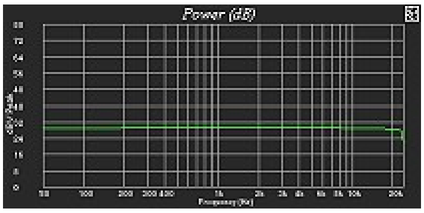
Apply Frequency Weighting (and Inverse) to I.R.

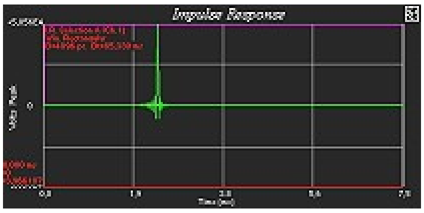
This function has the same behavior of the above Mic. Weighting function, but compensates the I.R. according to the currently selected frequency weighting (Settings/Compensation).

Apply Pink Filtering (and Inverse) to I.R.

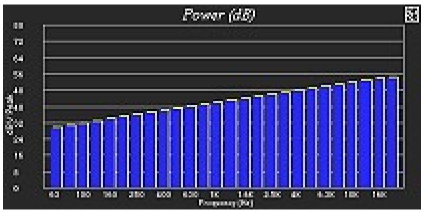
This function applies to the Impulse Response a Pink (or inverse Pink) filtering. This operation can be useful when analyzing in octave bands measurements performed with an unfiltered MLS signal.

Here is a pink filtering example:



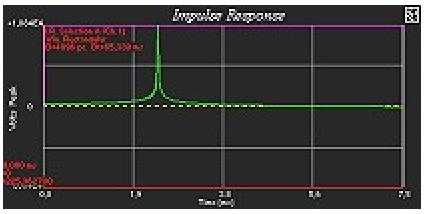


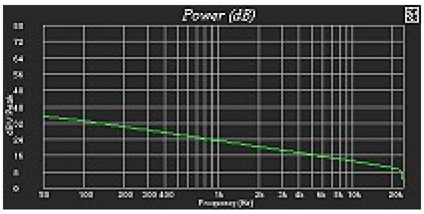
This is a simple loop-back measurement and gives a flat power spectrum when its frequency content is analyzed in narrow band. But when an 1/3 octave frequency analysis is required, the graph will look like the following:



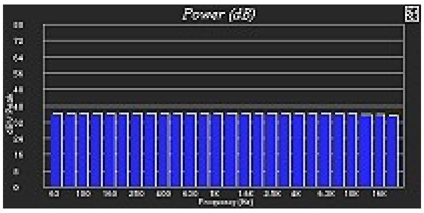
In the above plot, the effect of 1/3 octave band analysis is correctly shown: every subsequent band is about 1 dB greater than the previous one (3 dB in 1/1 octave analysis).

By applying pink filtering to the Impulse Response, the result will look as follows:





Now the 1/3 octave plot will look as follows:

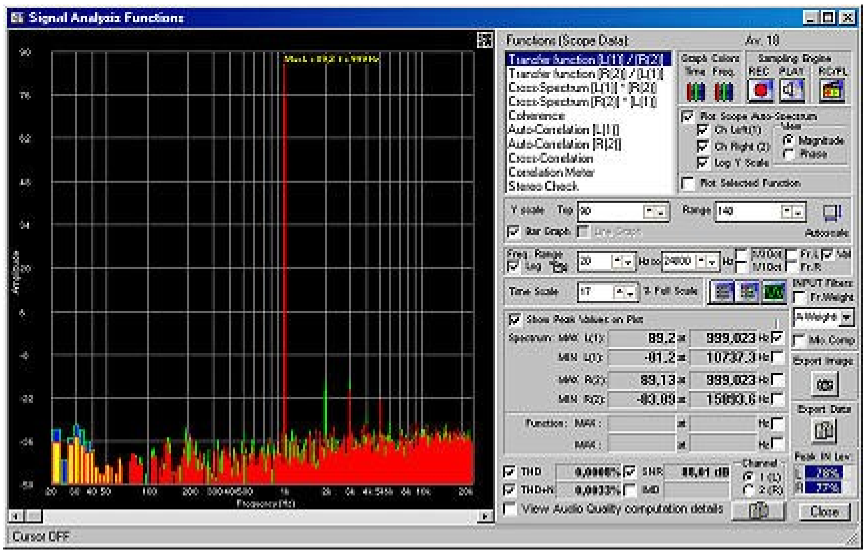


where every band has the same amplitude. See also Application note#6 about Pink Filtering.

Clear all Data Buffers

This function clears all data buffers (Time and Frequency) of the software.

RTA (Real Time Analysis)



In Sample Champion is possible to analyze simultaneously in real-time the 2 oscilloscope input channels and perform some DSP functions on them.

The RTA functions display can be opened by pressing the button:

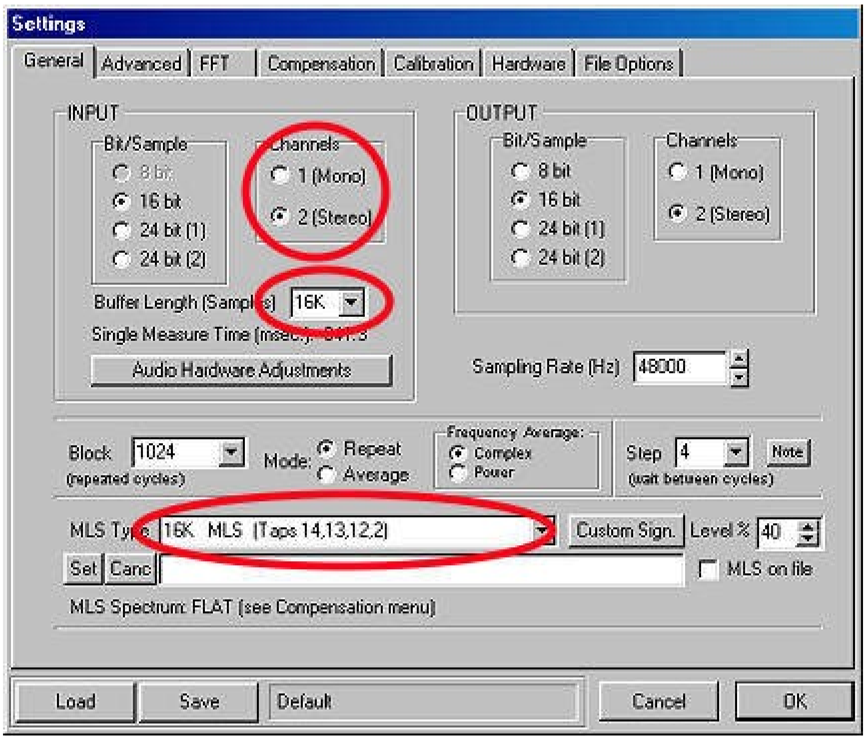
placed on the remote bar.

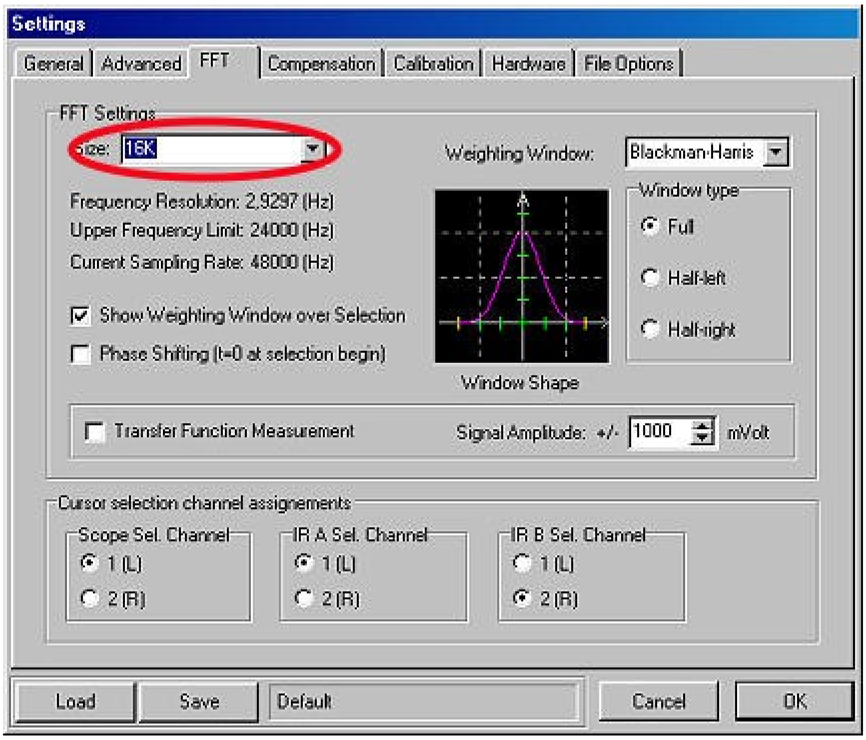
In order to operate properly, the RTA window requires that:

- Number of input channels = 2

- INPUT BUFFER LENGTH = FFT LENGTH

For example, for a buffer of **16K** samples, the settings are the following:

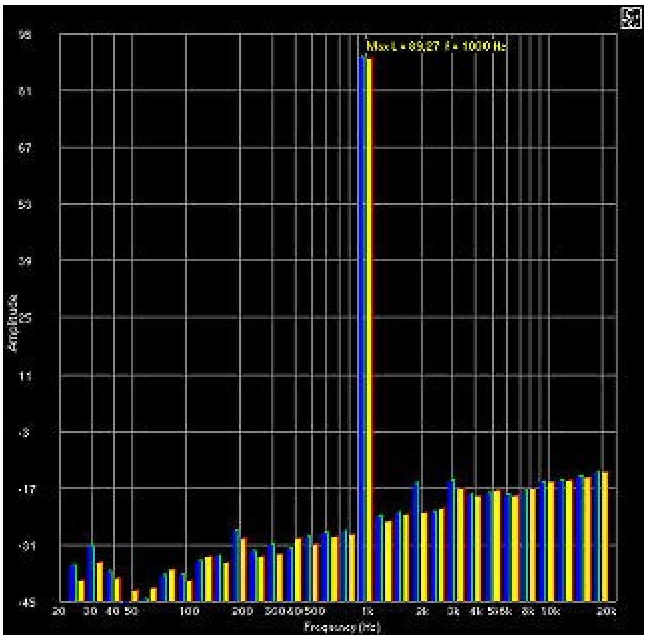


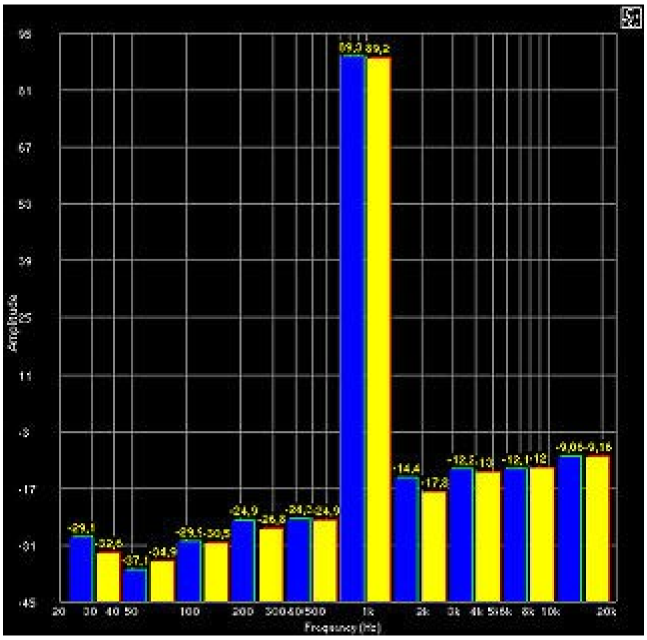


All the frequency windows in the main program must be closed when **RTA** functions are used.

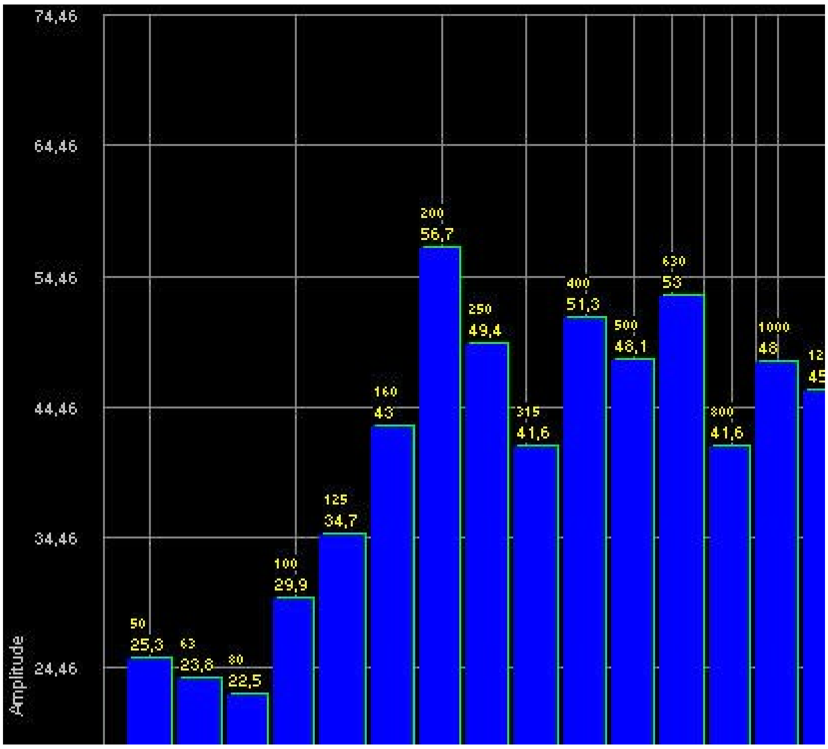
In the **SIGNAL ANALYSIS FUNCTIONS** window, the FFTs of one or both input channels can be plotted simultaneously, using different graph options. Note that, in some cases, when the FFTs of both channels are plotted, it could be necessary to use the line plotting option, to prevent the masking of one channel by the other.

Also 1/3 Octave and 1/1 Octave plot of both channels or of a single one are available:





It is possible to optionally write the Octave or 1/3 Octave value and frequency directly on the plot.



From the FFT data of the 2 input channels it's possible to perform in real-time and plot the following DSP functions:

- **Transfer function [L(1)] / [R(2)]** (Frequency Domain)

- **Transfer function [R(2)] / [L(1)]** (Frequency Domain)

- **Cross-Spectrum [L(1)] * [R(2)]** (Frequency Domain)

- **Cross-Spectrum [R(2)] * [L(1)]** (Frequency Domain)

- **Coherence** (Frequency Domain)

- **Auto-Correlation [L(1)]** (Time Domain)

- **Auto-Correlation [R(2)]** (Time Domain)

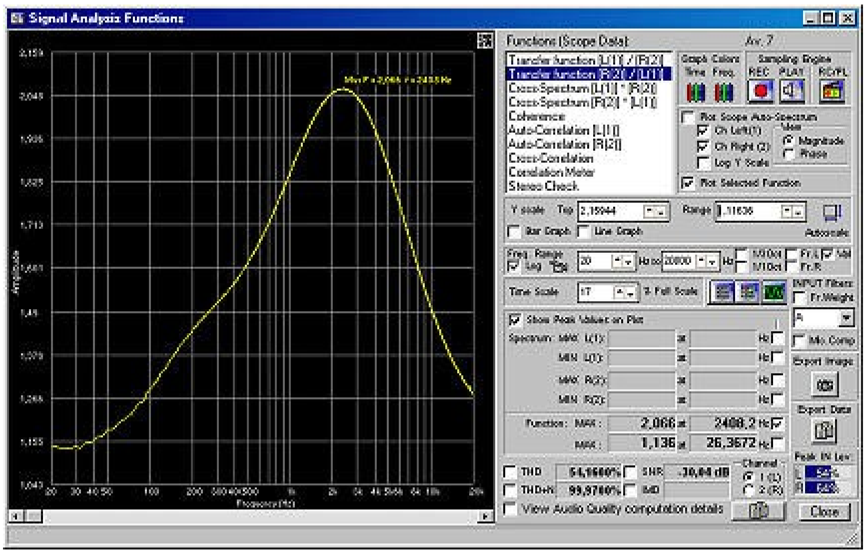
- **Cross-Correlation** (Time Domain)

- **Correlation Meter**

- **Stereo Check**

Transfer function

This function performs the complex FFT operation of channel 1 divided by the FFT of channel 2 (L/R) or the FFT of channel 2 divided by the FFT of channel 1 (R/L).

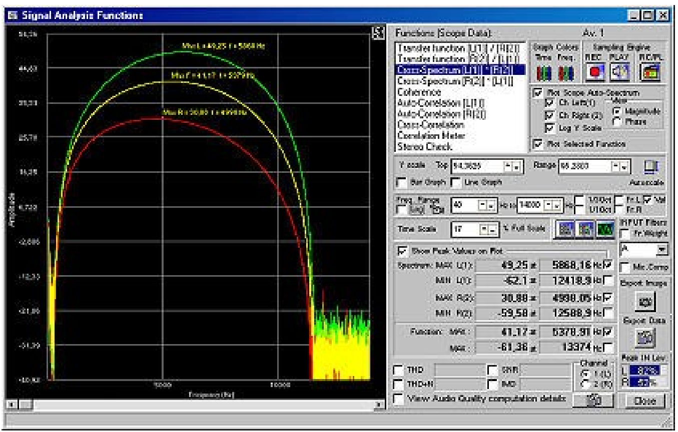


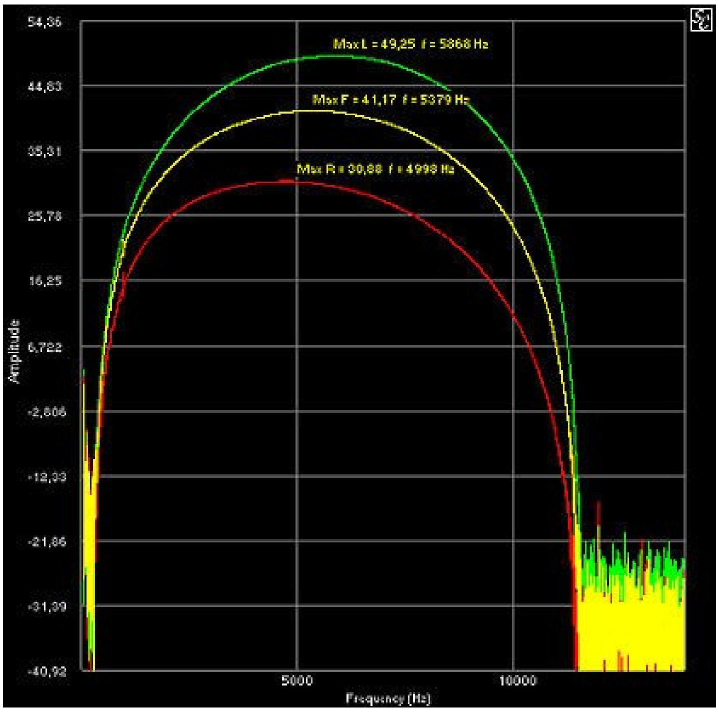
When measuring a transfer function, it's recommended to use a wide band signal (MLS) to obtain information on the entire audio band up to half the sample rate. In some cases it could be useful to make some averages to improve the measures.

IMPORTANT NOTE: when using the RTA functions, ONLY the **complex average** is available and NOT the power average. In other words, the average is done on the complex FFT data and when average option is enabled IT'S RECOMMENDED to **use ALWAYS the internal signal generator** to maintain the synchronism between each block of time data sampled and perform thus correctly the average.

Cross-Spectrum

This is computed by multiplying one complex spectrum by the complex conjugate of a second spectrum. It gives information about the power common to 2 signals.

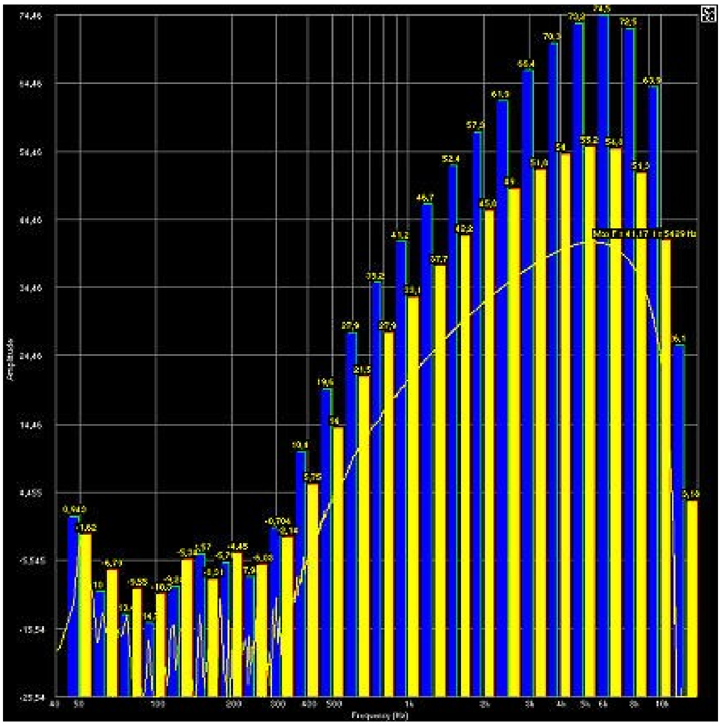




In the figure above a Cross-Spectrum example is shown. The spectrum of channel 1 is plotted in green, the spectrum of channel 2 in red and the Cross-Spectrum in yellow. The frequency axis is linear. Also the MAX values of the spectrums are shown on the plot .

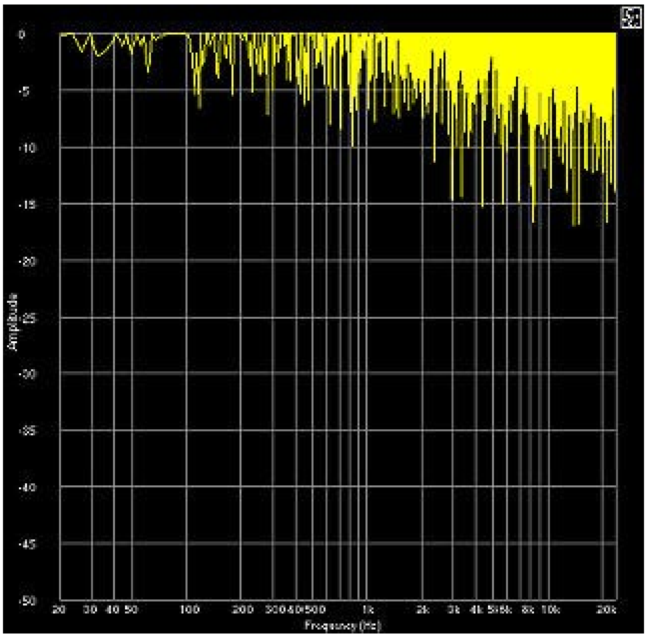
Note that the Cross-Spectrum of ch.1*ch.2 can be different from the Cross-Spectrum of ch.2*ch.1!

The spectra of the 2 input channels can be plotted on the same plot as the the selected function (in the frequency domain). The input spectrum can be plotted also in 1/3 octave mode, like in the example below (frequency axis logarithmic):



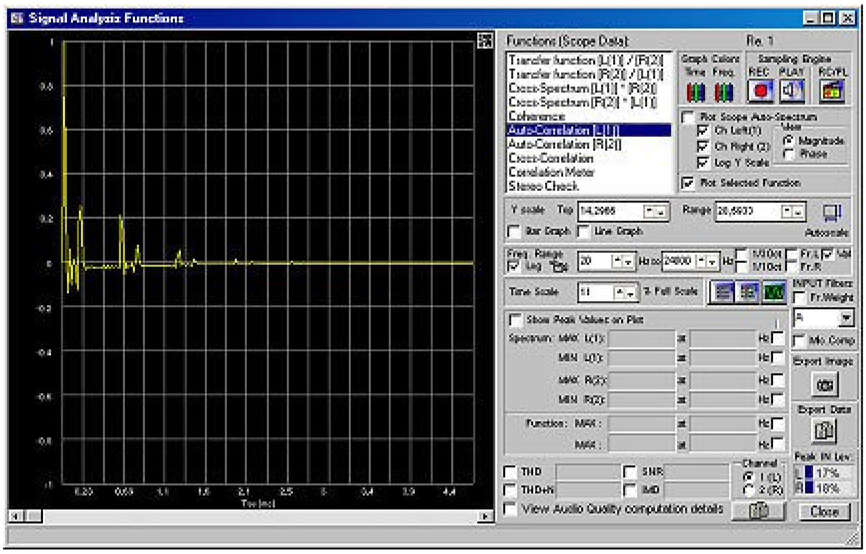
Coherence

This function is the ratio of the squared magnitude of the Cross-Spectrum and the spectrum of channel 1 multiplied by the spectrum of channel 2. It gives information about the mutual linearity of the channels.
The example below shows the coherence of a signal before (ch.1) and after (ch.2) a digital reverb effect.



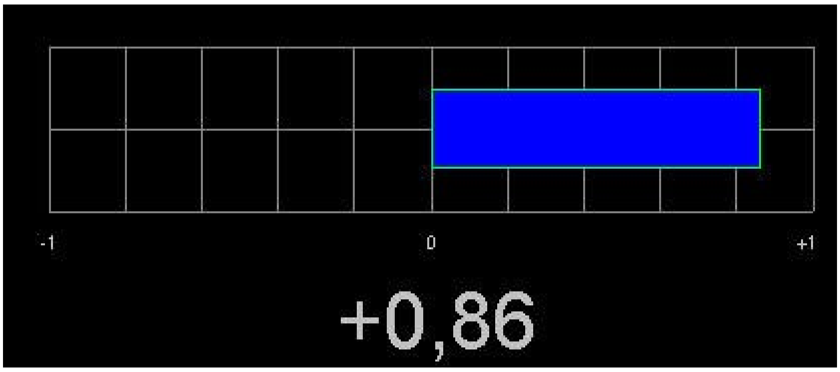
Cross-Correlation and Auto-Correlation

These Time domain functions give information about the mutual correlation of 2 signals (or the same signal in case of Auto-Correlation). It is very useful for finding, for example, echo or delays in time data. In the example below a signal has been passed through a digital echo effect; the peaks correspond to the delays (with feedback) set on the effect machine.



Correlation Meter

This function offers a quick and easy way to check the correlation between the 2 input channels. This measurement can be performed by pressing only REC (red) button (no signal generator) for measuring, for example, a stereo musical signal. In other cases it is possible to use the signal generator.



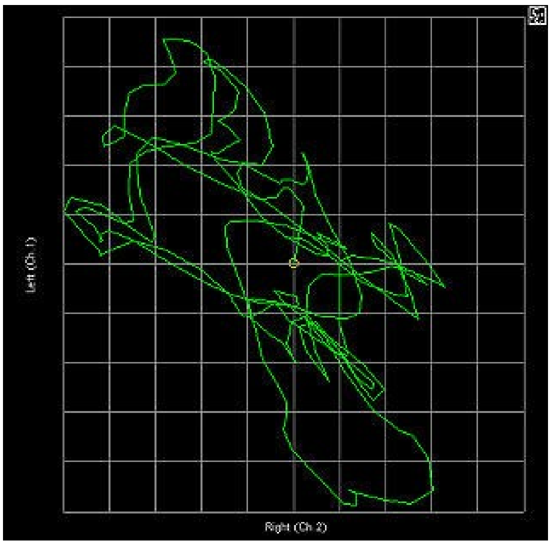
The result gives a correlation index between the 2 channels. Specifically:

+1 means that LEFT and RIGHT signals are identical
0 means that LEFT and RIGHT signals are not correlated
- 1 means that LEFT and RIGHT signals are identical and exactly out of phase (180º)

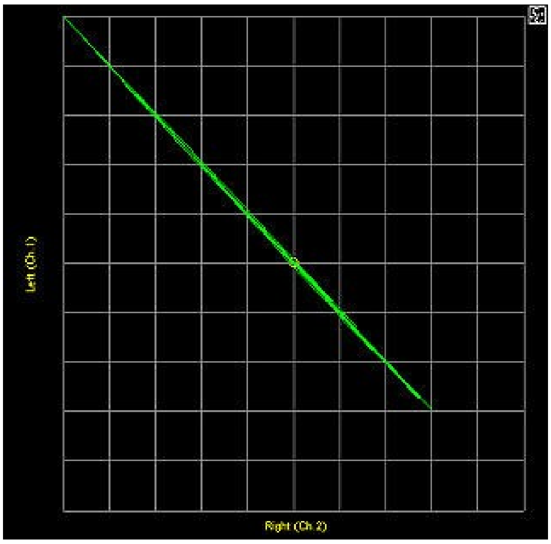
Stereo Check

This function plots the LEFT channel versus the RIGHT channel. It's easy, in this way, to obtain information about the 2 channels.

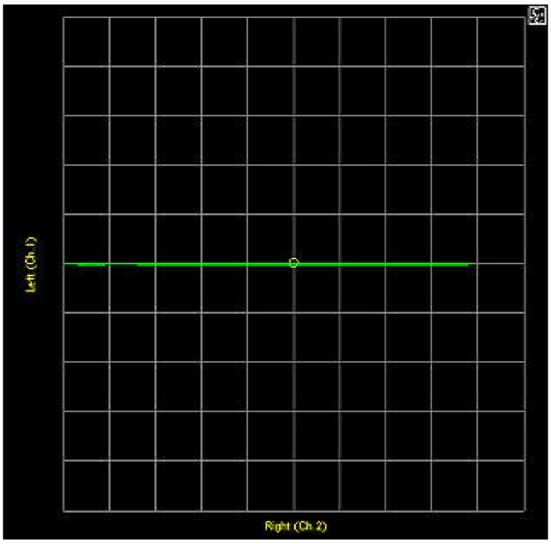
– Example of a stereo musical signal:



- Example of a mono musical signal. The amplitude of the 2 signals are exactly the same since the slope of the line is exactly 45 degrees:

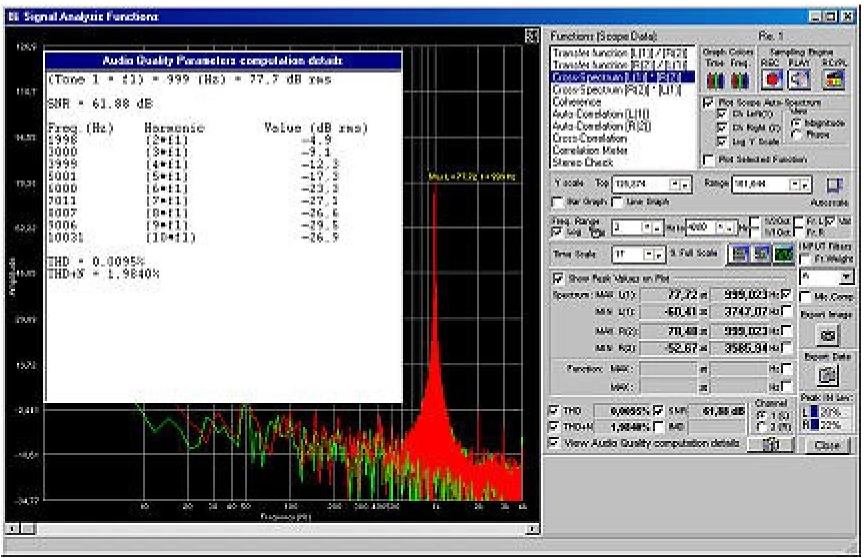


- Example of a signal present only on the right channel:



Audio Quality functions

In an early version of Sample Champion the computation of SNR, THD, THD+N and IMD had to be done in the Audio Quality Plugin. Now the computation of **SNR**, **THD**, **THD+N** and **IMD** can be performed in the RTA window.

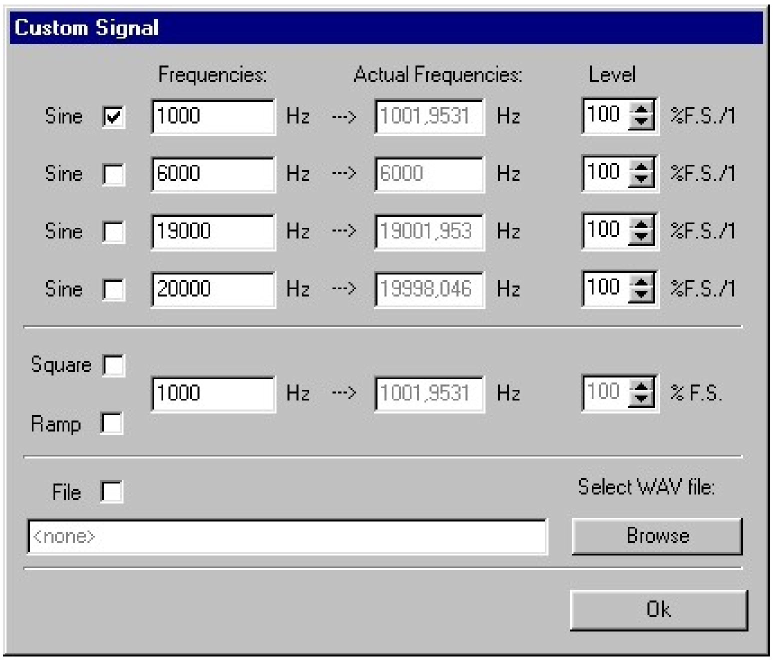


SNR (Signal to Noise Ratio) measurement

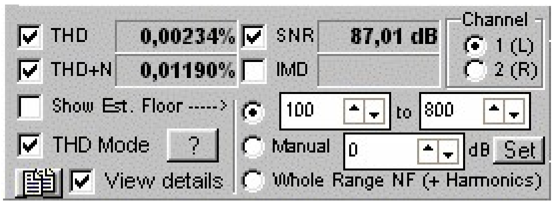
The **SNR** value is the ratio of the peak power level to the remainig noise power.

Measurement procedure:

In the **Custom Signal** Window (Settings/General/Custom Signal) a single pure tone must be selected. The following figure shows an example (1 kHz tone).



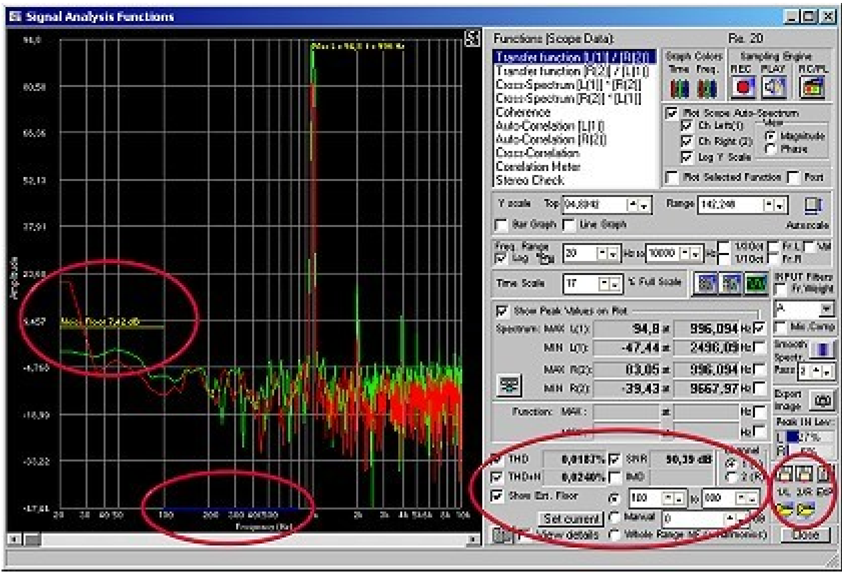
Then a SYNC REC/PLAY ( ) measurement cycle can be started (for example with a loopback connection).



If the Averaging Mode has been selected, at each cycle the SNR value will decrease until the minimum value is reached.

A data box containing all details about the parameters computation can be shown inside the data box (View details option).

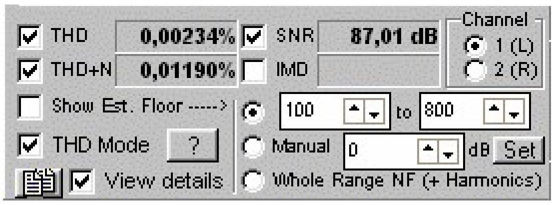
Different methods can be selected for the evaluation of the Noise Floor value in the SNR computation.



1- computation of the Noise Floor level as logarithmic sum of narrow band power levels in a **frequency range selected by the user**. This option can be used for estimating the Noise Floor WITHOUT the harmonics due to distortion; it is, for instance, possible to select the band 100..800 Hz when a 1 kHz tone is used as signal generator.

2- manual: this option allows to set manually the Noise Floor value. It can be measured, for example, recording the background noise in absence of input signals.

3- computation of the Noise Floor level as logarithmic sum of narrow band power levels on the **whole bandwidth from 20 to 20000 Hz**.



A yellow line corresponding to the Noise Floor level and a blue line under the frequency range selected by the user for the computation can be optionally visualized on the plot.

If the pure tone used for the measure falls inside the frequency range selected by the user, it is automatically excluded from the computation.

The Noise Floor value used for **THD+N** computation is not influenced by this option and is computed on the whole bandwidth; all harmonics generated by the 2 pure tones are also automatically excluded.

IMD computation is not influenced by this option.

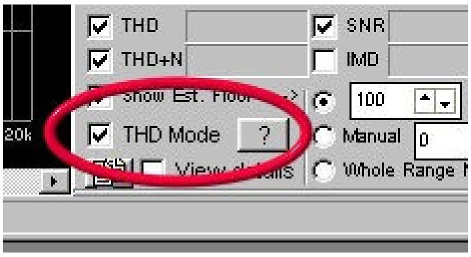
THD (Total Harmonic Distortion) measurement

THD and THD+N can be computed using in two different ways.

If the option **THD Mode** is unchecked, the following formulas are used:

$$\%THD = \frac{\sqrt{H_2^2 + H_3^2 + ... + H_N^2}}{\sqrt{H_1^2 + H_2^2 + H_3^2 + ... + H_N^2}} \times 100$$

$$\%THD+n = \frac{\sqrt{H_2^2 + H_3^2 + ... + H_N^2 + n^2}}{\sqrt{H_1^2 + H_2^2 + H_3^2 + ... + H_N^2 + n^2}} \times 100$$



If the option **THD Mode** is checked, the following formulas are used:

$$\%THD = \frac{\sqrt{H_2^2 + H_3^2 + ... + H_N^2}}{\sqrt{H_1^2}} \times 100$$

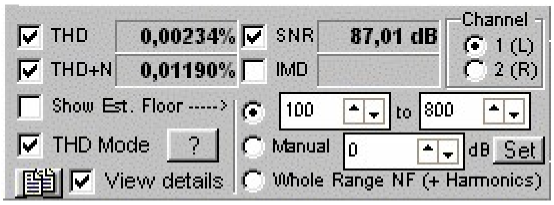
$$\%THD+n = \frac{\sqrt{H_2^2 + H_3^2 + ... + H_N^2 + n^2}}{\sqrt{H_1^2}} \times 100$$

In normal measurements (with low THD and THD+N) the 2 methods will give quite identical results, since more than 99% of the measured energy is always contained in the fundamental harmonic (H_1).

Measurement procedure:

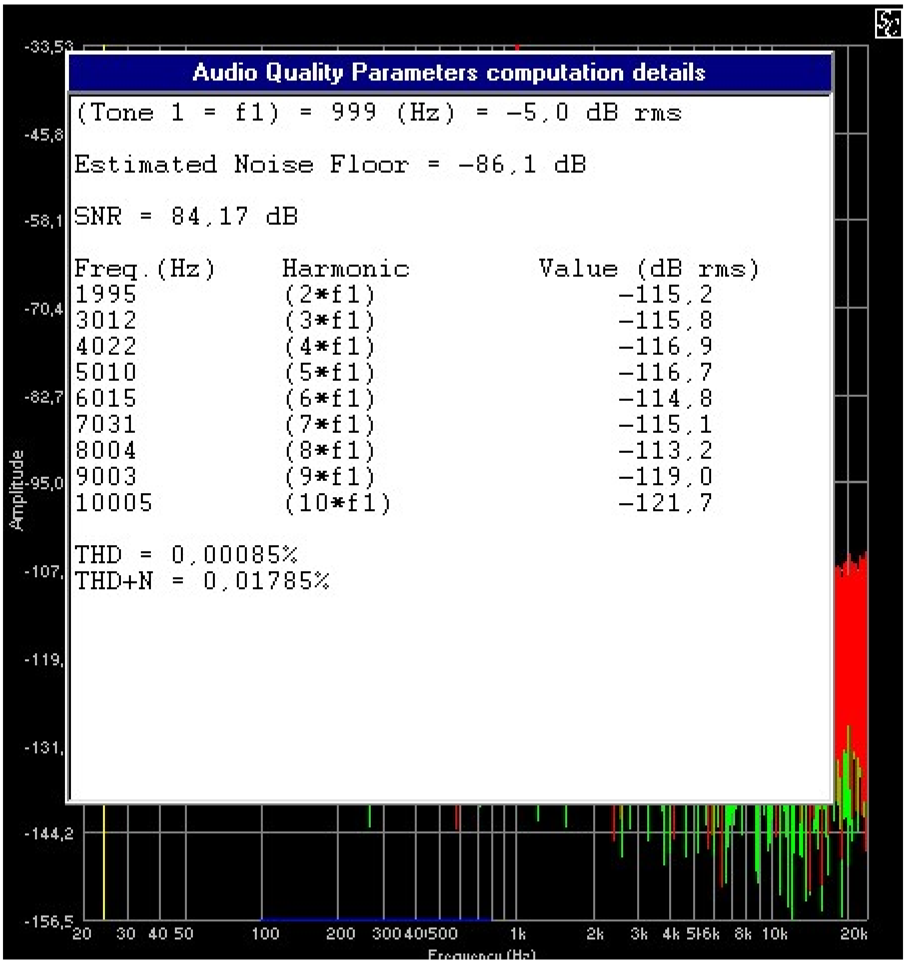
In the Custom Signal Window a single pure tone must be selected.

A SYNC REC/PLAY ( ) measurement cycle can be started.



THD and **THD+N** will be computed and shown.

The **View details** window can show additional information:



IMD (InterModulation Distortion) measurement

This parameter gives a measure of the distortion caused in the device under test by two pure tones (cross modulated power). The following harmonics are considered:

(Tone 1 = f1)
(Tone 2 = f2)
(f2-f1)
(f1-2*(f2-f1))
(f1-(f2-f1))
(f1+2*(f2-f1))
(f1+3*(f2-f1))
(2*f1)
(f1+f2)
(2*f2)
(3*f1)
(2*f1+f2)
(2*f2+f1)
(3*f2)

The **IMD** value is computed as the ratio of the sum of the power levels of the intermodulation harmonics to the sum of the power level of the two strongest tones.

Measurement procedure:

In the Custom Signal Window **two** pure tones must be selected.

Select the **IMD** function and start a SYNC REC/PLAY () measurement cycle.

Common choices of the two fundamental frequencies are:

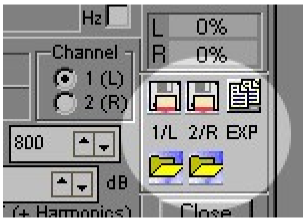
* **SMPTE**: 60 Hz and 7 kHz (4:1 ratio)
* **DIN**: 250 Hz and 8 kHz
* **CCIF**: 19 kHz and 20 kHz

SNR, THD, THD+N AND IMD MUST BE MEASURED ONLY USING THE INTERNAL SIGNAL GENERATOR

Load and Save Spectra in RTA window

Spectra measured in RTA window (narrow band, 1/1 or 1/3 octave) can be loaded and saved. The functions (Transfer function, Cross-Spectrum, Coherence, Auto-Correlation, Cross-Correlation) can be computed in post-processing mode on loaded data.

Left channel and right channel FFT banks can be saved and loaded independently.



Data of scope spectrum or functions can be exported as TXT files (with options set in Settings/File Options) and the graph can be saved as BMP image.

POINTS TO REMEMBER WHEN USING RTA FUNCTIONS:

- THE AVERAGE IS COMPUTED ONLY IN COMPLEX MODE

- ONLY THE INTERNAL SIGNAL GENERATOR MUST BE USED

- IF THE YELLOW LABEL "BUFFER OK" ON THE REMOTE BAR IS NOT VISIBLE EVERY SAMPLING CYCLE, THE STEP VALUE MUST BE INCREASED TO AVOID LOSS OF DATA

Troubleshooting – F.A.Q.

[1] Which is the accuracy of a measurement done with Sample Champion ?

The accuracy of a measurement (not considering external devices like microphones, amplifiers...) depends basically on the sound card. You can evaluate the performance of your sound card using Sample Champion itself in a loop back configuration. See application notes and FAQ #4.
If your sound card has a poor frequency response, some improvements could be obtained using MLS filtering and the equalization capabilities of Sample Champion. A nice solution could consist in using an audio card with S/PDIF digital IN/OUT and an external high quality A/D-D/A converter.

[2] Which sound cards are more suitable for Sample Champion?

Any sound card working under windows 95/98/2000/NT4 in full-duplex mode is suitable. Nowadays all sound cards satisfy these requirements.
Some sound cards have peculiar characteristics.
For example:

Turtle Beach audio cards (Pinnacle, Fiji...) have an excellent frequency response, low noise and in our tests **NEVER** lost a bit of sound data or the sync between input and output audio streams. They work very well also on slow computers because of their hardware architecture and efficient software drivers. These audio cards are **recommended** for Sample Champion.

Last generation SoundBlaster cards (ex. SBLive, PCI128...) have a very good frequency response but can be used with Sample Champion **ONLY** at the sampling frequency of 12 kHz, 24 kHz and 48 kHz because they work internally always at 48 kHz. Other sampling frequencies are obtained by an internal frequency conversion that can give poor results with Sample Champion.

Good old first generation SoundBlasters (SB16, SB32, some AWE...) used on slow computers can experience sync losses caused by software interference with the video cards. Sample Champion, in *Blinded Mode*, performs a workaround. Anyway we reccomend to update your sound card!

[3] Why Sample Champion requires a lot of RAM memory?

This is a consequence of the very high accuracy used for all internal operations. For example the FFT (limited for the user to 64K, but extended for some internal operations to 256K) is computed on double-precision complex data and using both frequency bands (positive and negative frequencies) to perform some specific signal processing operations and in prevision of any future DSP expansion (using special purpose plugins). To prevent any resizing or reallocating of the virtual memory during measurements, Sample Champion allocates, once started, all RAM memory required for working.

[4] How can I test whether Sample Champion works perfectly with my PC configuration?

First of all **download the full-working trial version** and install it. The main test to be performed is whether your computer is fast enough to perform the real-time operations required by Sample Champion and nothing in the system interferes with the input and output audio streams.

The test:

- Connect in loop back the input and output of your sound card
- Start Sample Champion and open the Impulse Response and Info windows
- Set a sampling frequency of 44100 Hz and the 'Repeat' mode
- Start a Sync Rec&Play loop of 4096 cycles and read the peak position of each cycle, in the info window
- Adjust, if necessary, the in and out levels from the mixer to obtaind a distorsion-free impulse response; increase the 'Step' value (if required, see the manual for details) until you can read in the control bar the yellow label 'BUFFER OK' for at least 1 second. Repeat the above procedure several times.

If the peak is at the same position at **every** cycle, your computer can perform real measurements with Sample Champion (once calibrated, the peak will be near t=0).

- **Note 1**: When a measurement cycle starts, it is normal to observe a small change of the peak position, that could be advanced or delayed by one or two samples with respect to other measurement cycles. The peak must, however, remain at the same position during a measurement cycle.

- **Note 2**: See the FAQ about sound cards. If you have a SBLive or similar, perform the test at 48 kHz.

- **Note 3**: If the peak changes its position, during the test, this means that something interferes with either input or output audio streams. This is not due to Sample Champion but to the specific software/hardware configuration. **WARNING:** If this happens, **all** programs manipulating audio data in real-time (for example multitrack recording or other measurement software) on your system could experience the same problems. However only performing synchronous playing and recording (and expecially using correlation techniques with MLS) it is possible to discover them! Sample Champion tells you the truth!

ANYWAY, the above test is very significant and, if performed correctly with positive results, can assure you that Sample Champion works fine.

If the problem is caused by the sound card, see the above FAQs about most suitable sound cards.

[5] About full-duplex operations, should I play at 8, 16 or 24 bit the MLS signal?

Sample Champion can play at 8, 16 or 24 bit resolution (8 for backward compatibility with some old audio devices). When playing tonal signals it is recommended to use at least 16 bit resolution, but when playing MLS signals (just a series of +K and -K, where K is the output value) it doesn't matter. In our tests we didn't find any difference using 8, 16 or 24 bits in playing MLS signals.

[6] My SoundBlaster audio card refuses to perform full-duplex operations

Download and install the latest driver! Remember also to disable 3D spazialization enhancements and set bass and treble controls to zero when performing measurements. If any loss of sync is observed (see FAQ #4) try also other IRQ and DMA settings because Windows can share some of these channels and some configurations could give poor performances in terms of audio quality.

Troubleshooting – TIPS

[1] How can I optimize my operating system for real-time audio measurements?

You should eliminate all not strictly necessary programs running in background. In Windows 95/98 by pressing CTRL-ALT-DEL it is possible to see all running tasks (at least those installed by the user). Close all programs and windows then press CTRL-ALT-DEL: the best thing would be to see listed only "Explorer". In Windows NT4 or Windows 2000 use the taskmanager to see all running processes and try to close all unnecessary ones. Refer to the operating system manual to avoid automatic start of the unnecessary tasks.

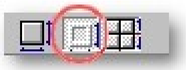
[2] How avoiding common errors using digital audio cards and improving sampling result?

The worst thing that can happen working with digital audio is distorsion caused by excessive input levels. For musical audio recording this is a big problem because distorsion caused by A/D saturation is usually very unpleasant. The same problem can occur when making measurements with digital instrumentation.
The suggestion is to monitor constantly the peak level meters and adjust consequently the input level in the mixer. As a rule, the maximum peak level should be at least 6 dB under the saturation level.
Note that in some old sound cards the output section (D/A) can give some distorsion on the last bit (actually playing signals higher than 16384=2^14 in absolute value). If you notice this using Sample Champion, the solution is simple: in Settings|Level use

values<50%. Fortunately Sample Champion gives you a simple way to analyze the acoustical performance of your audio card in terms of distorsion in order to perform a fine tuning of your level settings. See the application notes concerning this kind of measurements.

[3] In Sample Champion the wave and the spectrum data seem to "jump" high and low. What happens?

Sample Champion has the "Y Auto Scale" option turned on (default).



Simply perform a test measure to let the program measure the maximum peak amplitude and then turn off the autoscale or, in a repeated or averaged measure cycle, turn off the option once that the result of the first measure is plotted. When the austoscale is off, it is always possible to perform a manual autoscale by pressing the button on the left of the autoscale button.

Troubleshooting – Known issues

- We had a problem sampling with **TurlteBeach Pinnacle** and length of the input buffer equal to **16K samples** (with pure tones or custom waves, **NOT** MLS signal). The problem is certainly due to a bug in the Pinnacle drivers, since it is not present with other audio cards.
Solution: if you experience this problem, avoid that input length (use instead 8K or 32K).
- This problem does not affect MLS measurements (impulse response) because when a 16K MLS signal is selected, the actual input length is 16K-1 samples.

- **Windows 2000** seems to be very stable and suitable for audio measurement software. But some precautions must be taken:
- We found that W2K does not "inform" the audio software of the correct **maximal sampling frequency** allowable for a specific sound card. The consequence is that, when a sampling frequency not allowable is selected, the system could hang (in this case try stopping the non responding application with Task Manager). This affects all audio software and not only Sample Champion.
Solution: see your sound card manual to learn the maximal allowable sampling frequency.
- Another minor problem we found concerns the **audio mixer** of W2000: for some audio devices, the "Enable"-"Disable" checkboxes are inverted.
Solution: with Sample Champion, make some test measurements and check the input level meters to see if the signal IN/OUT are correctly selected.

- Troubles have been noticed during the concomitant execution of Sample Champion and other programs when a reduced disk space (less than about 100 MB) is available. This

happens because Windows creates a swap file (Win386.swp, usually located inside the Windows directory) for emulating the RAM memory required by specific programs (when not present as RAM). The disk space required by Sample Champion (when all RAM memory is used by other software) is about 70 MB or more, depending on settings and opened plugins. If this amount of free disk space is not available, Sample Champion will not work properly and will terminate with unexpected errors. The symptoms are error messages such as **Plugin_default not found** and other fatal errors during startup. **Solution:** Leave at least 200 MB of free space on the disk where the Windows swap file is allocated. Close **any** other software before starting Sample Champion (this is always recommended).

Troubleshooting – Most common problems (and solutions)

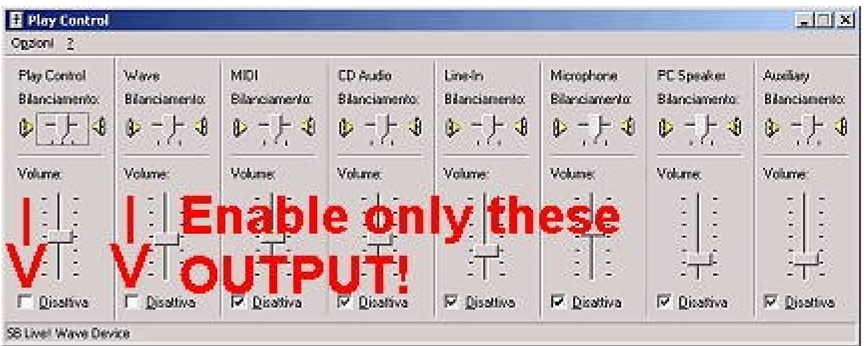
The first configuration to be performed before making any test measurement concerns the audio card mixer. In the following examples a Sound Blaster SB Live! audio card is considered, but the suggestions are valid also for any other device. The operating system was Windows 2000.

1- Connect **Line IN** to **Line OUT** with a jack-jack cable and start Sample Champion. Open the **Impulse Response Window**.

2- Close any other software, including any taskbar application.

3- Open the internal Sample Champion mixer or the standard Windows mixer and enable ONLY the following:

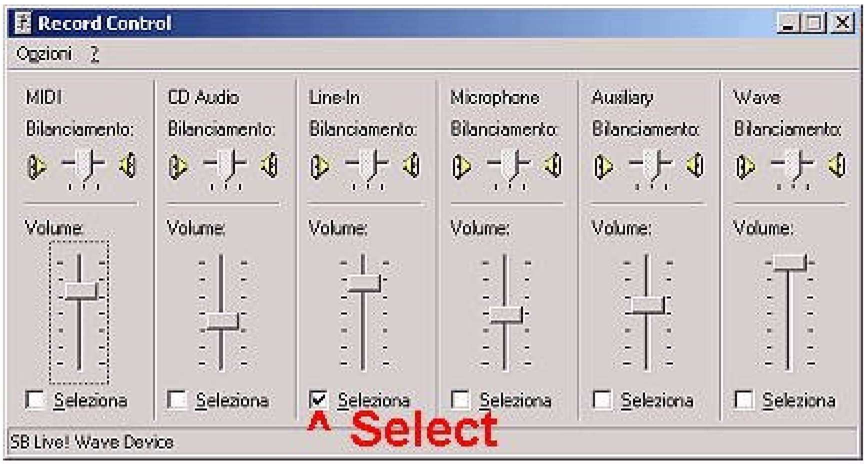
PLAY Control:
- Enable **Play** (set level to middle)
- Enable **Wave** (set level to middle)
- Disable all others



Play Control settings (SB Live!)

RECORD Control:

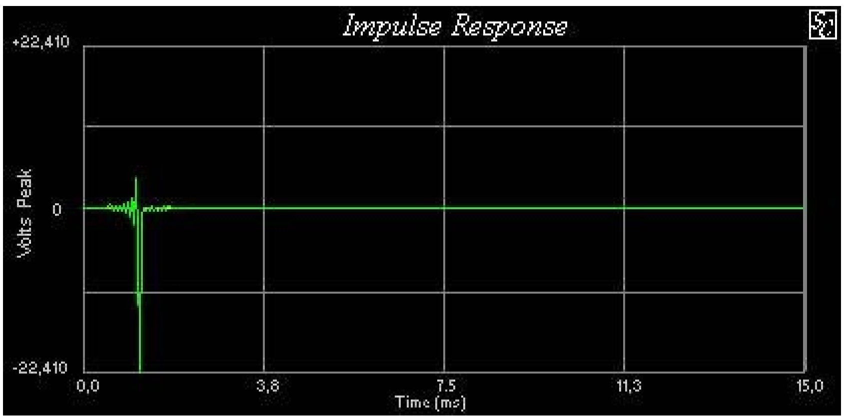
Enable **Line-In** (set level to middle)
Disable all others



Record Control settings (SB Live!)

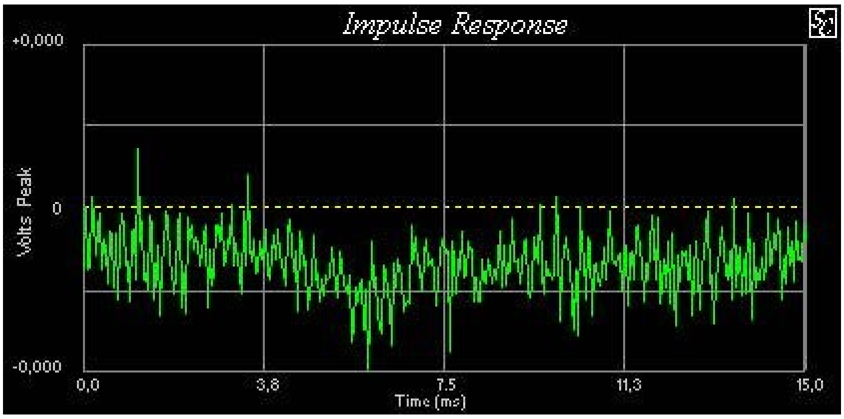
In Settings/General, set:
- 1 Channel In
- 1 Channel Out (16 bit)
- Buffer Length 4K
- Block = 16
- Mode = Repeat
- Sampling Rate = 48000 Hz
- Step = 4
- MLS type = 4K
- Level = 50%

4- Now press **Syncro Start/Stop MLS and Sampling** button and look at what happens.



Correct Loop-Back Impulse Response

If you will see a still Impulse Response like in the above figure, congratulations, you are lucky! Sample Champion is running correctly and your hardware/software configuration is suitable for MLS measurements. Anyway some other tests with different parameters are suggested for an extensive test.



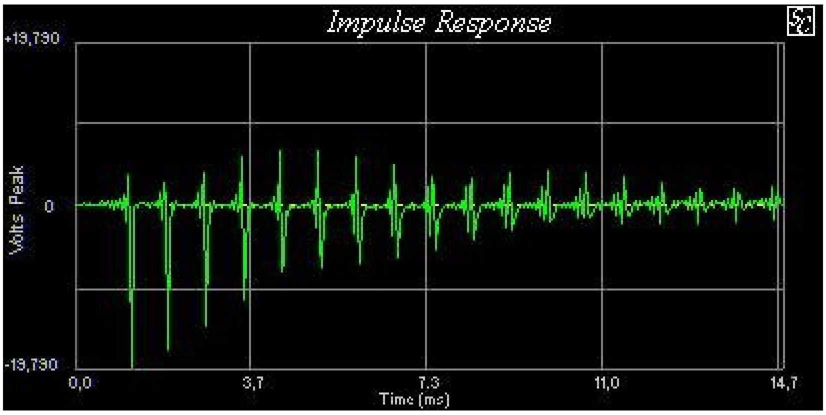
Absence of signal Loop-Back Impulse Response

If the computed Impulse Response looks like in the above figure, this means that the chain between input and output is interrupted. Check the loop-back connection and mixer settings.
The above plot can be obtained also by pressing the **Rec** button without any MLS signal in output; it is due, in this case, to the cross-correlation between the selected MLS signal (Settings/General, MLS Type) and (uncorrelated) background noise.

Most common problems

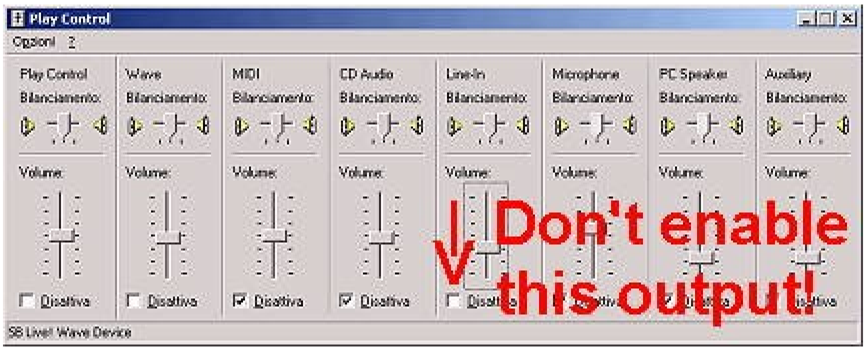
Echo in the measured Impulse Response

If the measured Impulse Response (?!) looks like the following



Wrong Loop-Back Impulse Response

most probably the cause is a wrong mixer setting. In the above example a repetition of the first (correct) peak can be easily observed. In this case the **Line-In** volume in the Play Control of the mixer had been erroneously enabled.



Wrong Mixer Setting

In other audio devices there are other similar level settings that **must** be disabled to obtain correct measurements. For example in the Pinnacle audio card mixer there is an "Input Monitor" control: set it to zero (and, if present, check also the mute button).
Obviously every "3D sound", "Reverb", "Chorus", "Spatialization" or other similar controls **must** be disabled for these kinds of measures.

Not still Impulse Response

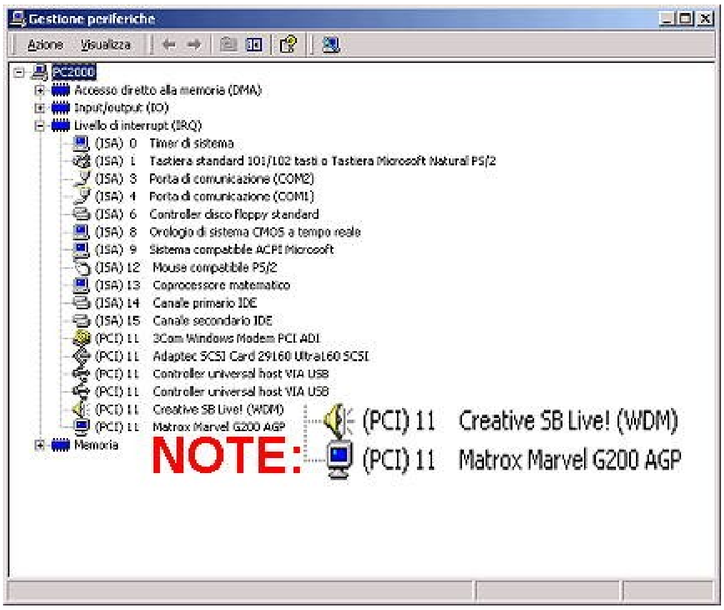
When the measured Impulse Response has a correct shape for a loop-back measure, but does not remain still during the cycle measures, the reasons could be different:

1- The sound card works internally at 48 kHz and emulates other sampling frequencies (this is the case, for example, of the SB Live). In this case, for MLS measurements, only 48, 24 and 12 kHz sampling frequencies **must** be used. Other frequencies will produce an unclean and not still Impulse Response.

2 - The sound card shares an IRQ with other devices of your system.
NOTE: this affects every audio software working in your system, so it could be worth to avoid it, even if you will not use Sample Champion.

The worst situation concerns IRQ sharing between audio and video devices because during sampling graphic information is often shown and this can cause problems.

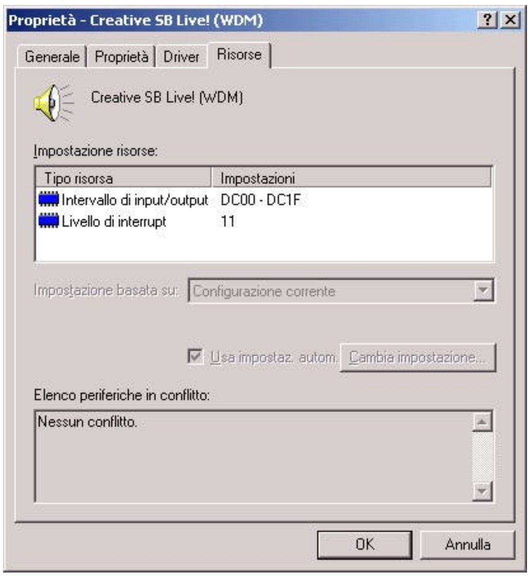
This is an example of IRQ sharing (to be avoided!):



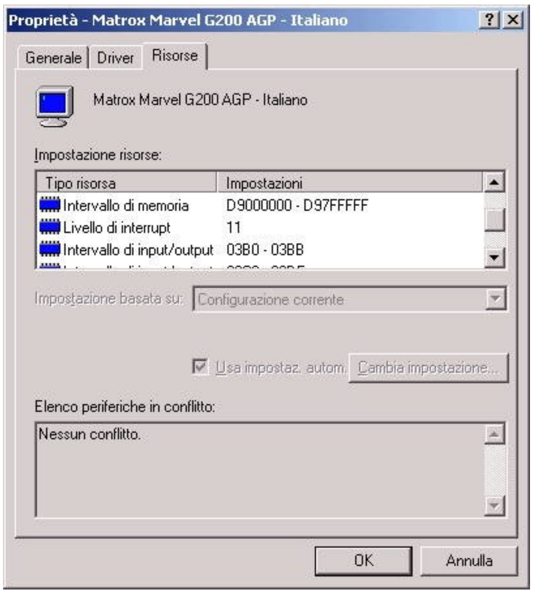
System configuration IRQ resources

As shown above, IRQ 11 is shared between Modem, SCSI adapter, USB adapters, audio card and video card!!
By the way, IRQs 2, 5, 7 and 10 are free.
The solution is to assign to the audio device one of the free IRQs. Sometimes uninstalling and reinstalling the device could be necessary, but it will be surely worth!

The figures that follow show in detail the settings of the audio and of the video cards.



Audio Card IRQ setting

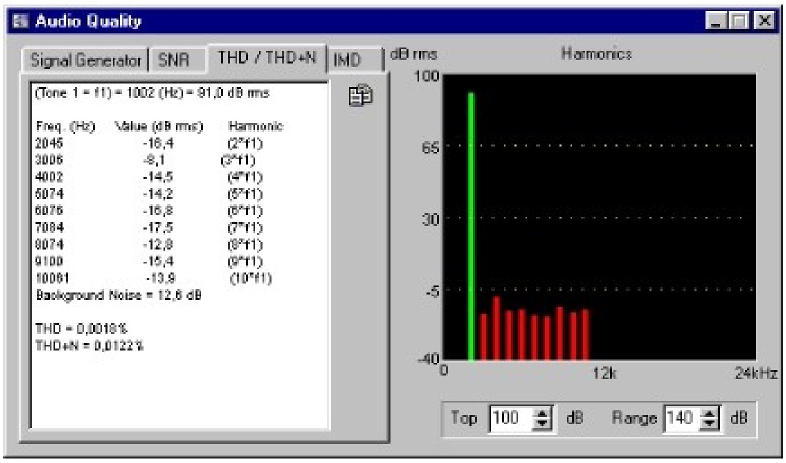


Video Card IRQ setting

In this case Windows doesn't allow the manual configuration of the audio device IRQ and the reinstallation could be necessary. For this purpose, follow the audio device and Windows installation guides.

Other features (available in the plugins)

Audio Quality Plugin



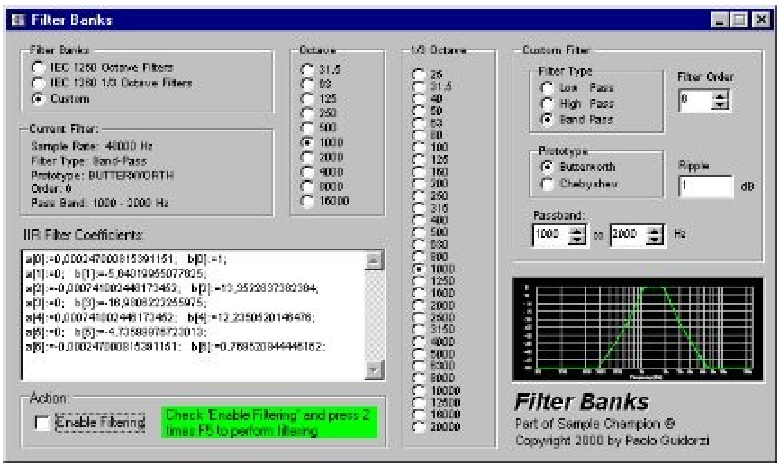
This module performs the following measurements:

- **SNR** (Signal to Noise Ratio)
- **THD** and **THD+N** (Total Harmonic distortion)
- **IMD** (Intermodulation Distortion)

More information and instructions about using the plugin can be found in the **Application Note #8**.

NOTE: Audio Quality Plugin is still available but discontinued. All its functions (with more features) are now included inside the RTA window!

Filter Banks Plugin (freeware)

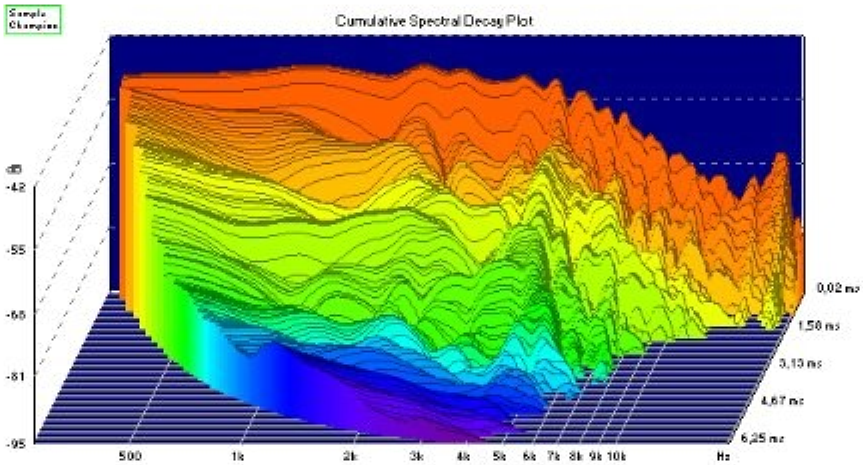


This module allows to approximate BUTTERWORTH and CHEBYSHEV low-pass, band-pass and high-pass analog filters by means of IIR (Infinite Impulse Response) digital filters. Moreover two banks of IEC 1260 Octave and 1/3 Octave Filter Banks are implemented as presets. This module can be used to filter a measured or loaded impulse response.

- The computed coefficients of the IIR filter can be exported to clipboard.
- The plugin is included in the setup package.

More information and instructions about using the plugin can be found in the **Application Note #9**.

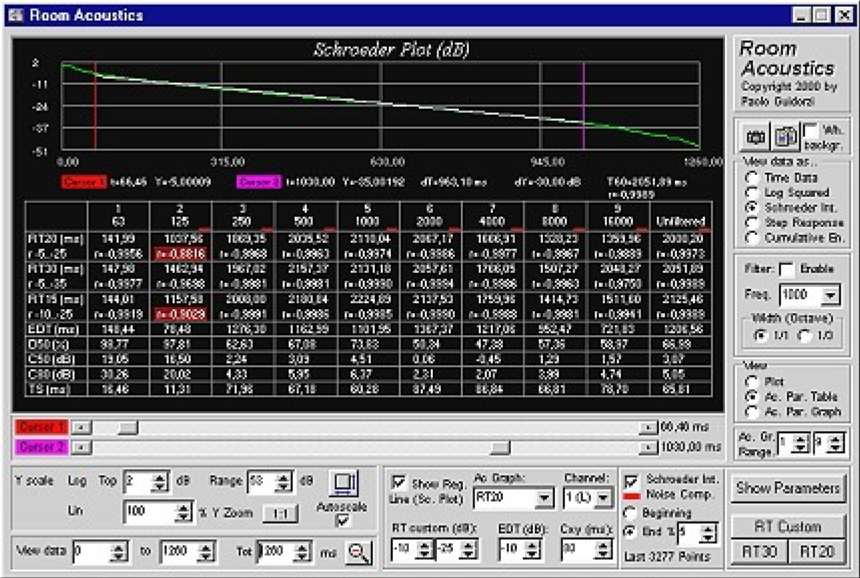
Waterfall Plot Plugin



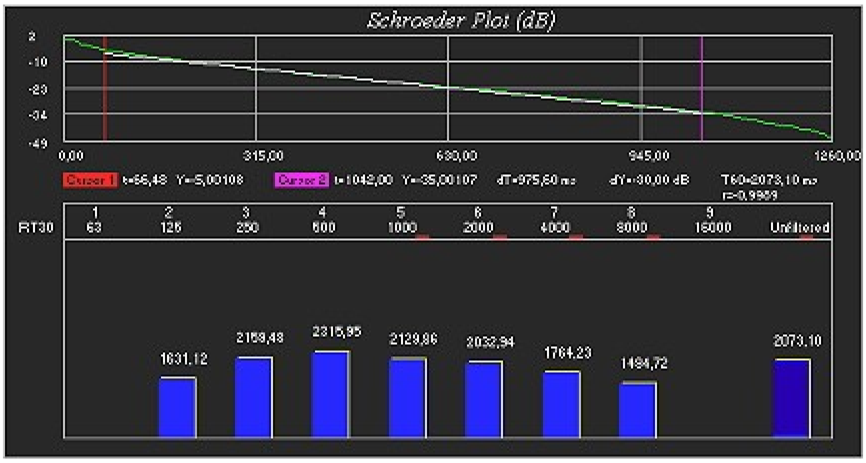
This module computes and plots (see the above figure) a **3D Cumulative Spectral Decay** of an impulse response. This is an invaluable information for finding loudspeaker and box resonances. Many options are available, such as different FFT sizes and weighting windows (independently from the main program settings). Unreliable computed low frequencies are automatically rejected from the plot.

More information and instructions about using the plugin can be found in the **Application Note #10**.

Room Acoustics Plugin

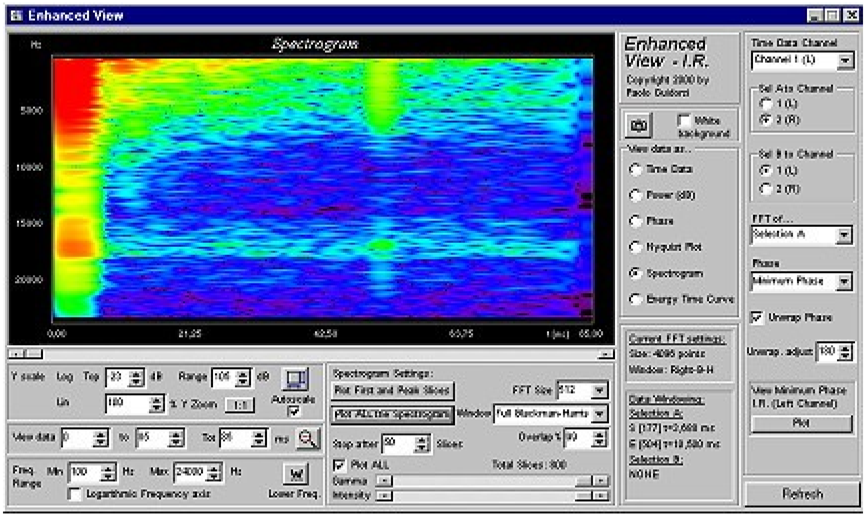


This module computes and plots many acousical parameters, such as **Reverberation Time**, **Early Decay Time**, **Definition**, **Clarity** and **Centre Time**. Many options and features are available: the acoustical parameters can be computed in wideband or in octave or 1/3 octave bands. The results are available in tabular or graphic form.
Time data can be plotted in many modes: **Log Squared**, **Schroeder Plot**, **Step Response**, **Cumulative Energy**. **Inverse Schroeder Integration** can take advantage of an user selectable **noise compensation**. The computed data can be exported to Clipboard and the plots can be saved on disk as Bitmap images.

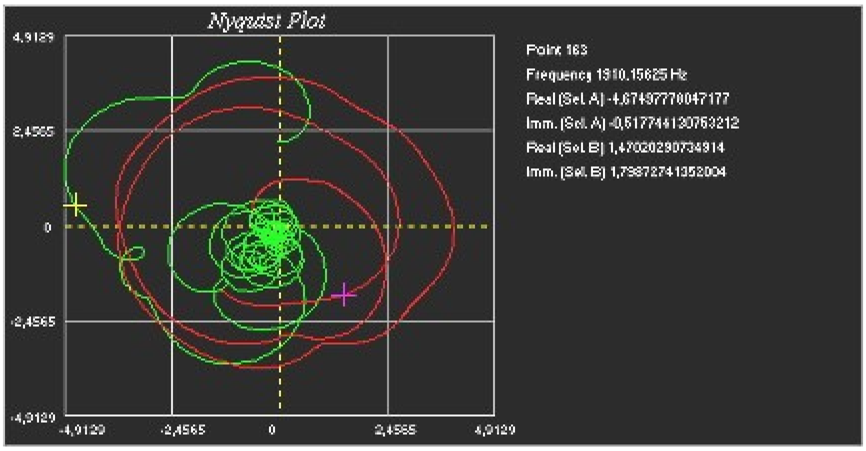


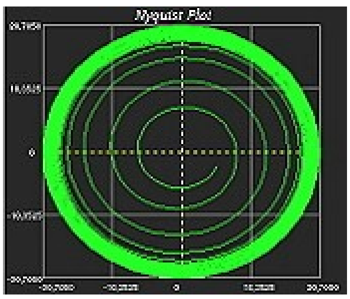
More information and instructions about using this plugin can be found in **Application Note #11**, **Application Note #12**, **Application Note #13** and **Application Note #14**.

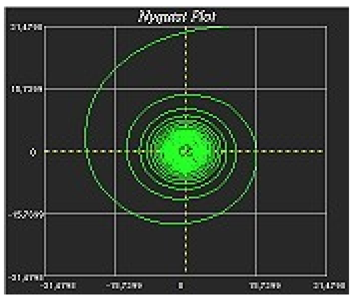
Enhanced View Plugin (freeware)

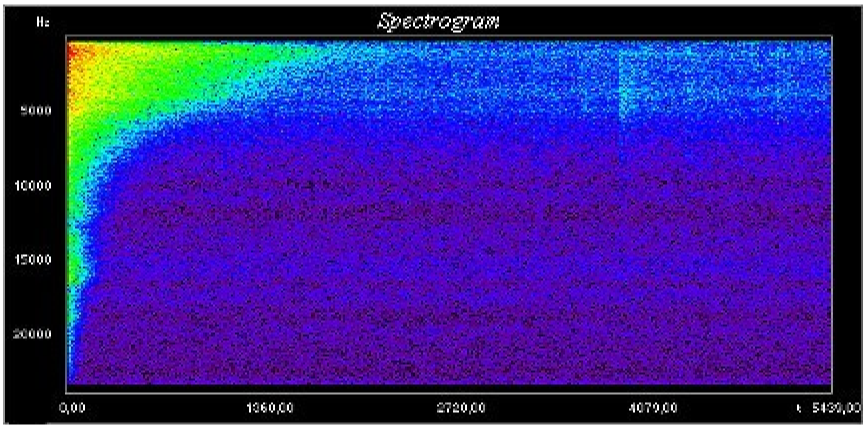


This module extends the Impulse Response view modes and analysis. It can compute and plot **Energy Time Curve**, **Spectrogram**, **Nyquist Plot**, **Phase Unwrapping**, **Minimum** and **Excess Phase** and **Minimum Phase Impulse Response Plot**.



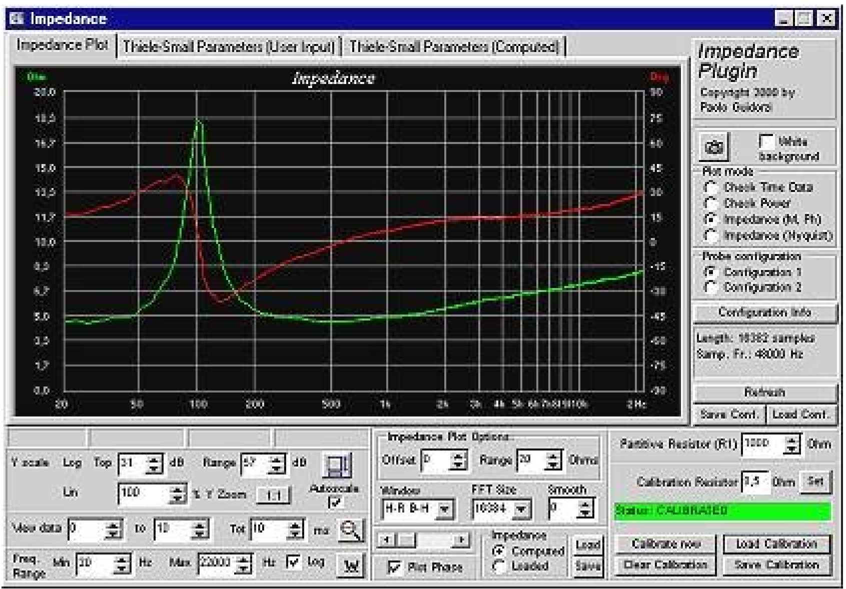






More information and instructions about using the Enhanced View plugin can be found in the **Application Note #15**.

Impedance Plugin



This module allows measuring Loudspeaker impedance and the computation of Thiele-Small parameters.

More information and instructions about using this plugin can be found in the **Application Note #16**.

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http://www.purebits.com
info@purebits.com